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SEPTEMBER 1981
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Publisher
Larry Zide

Editor
John M. Woram

Associate Editor
Mark B. Waldstein

Advertising Production & Layout
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Classified Advertising
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Circulation Manager
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Art Director
Bob Laurie

Graphics
K & S Graphics

Typography
Spartan Phototype Co.

sales offices

Roy McDonald Associates, Inc.
Dallas, Texas 75237
First Continental Bank Bldg.
5801 Marvin D. Love Freeway
Suite 303
(214) 941-4461

Denver, Colorado Area
Englewood, Colorado 80112
14 Inverness Dr. East
Bldg. 1—Penthouse
(303) 771-8181

Houston, Texas 77036
6901 Corporate Drive
Suite 210
(713) 988-5005

Los Angeles Area
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424 West Colorado Street
Suite 201
(213) 244-8176

Portland Area
Hillsboro, Oregon 97123
510 South First
P.O. Box 696
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San Francisco Area
Emeryville, California 94608
265 Baybridge Office Plaza
5801 Christie Avenue
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DEPARTMENTS

FEATURES

ABOUT THE COVER

• Seated at the controls of the WABC Production Control Room in New York City is Debra Iacovelli, staff engineer, WABC/WPLJ Radio.

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Calendar

SEPTEMBER

25 **The Society of Broadcast Engineers 9th Annual Central New York Regional Convention.** Syracuse Hilton Inn. For more information contact: Convention Chairman Hugh Cleland, WCNY TV/FM, Liverpool, NY 13088. Tel: (315) 457-0040.

OCTOBER

7-9 **Natural Stereo Techniques for Recording Music Workshop.** University of Wisconsin at Eau Claire. For more information contact: Burton Spangler, Audio Coordinator, Media Development Center, UW—Eau Claire, WI 54701. Tel: (715) 836-2651.

13-15 **The 11th Conference of the Western Educational Society for Telecommunications.** Harrah's, Reno, Nevada. For more information contact: Dr. Donald Price, Media Production Services, California State University, Los Angeles, CA 90032. Tel: (213) 224-3396.

25-30 **The 123rd SMPTE Technical Conference Exhibit.** Century Plaza Hotel, Los Angeles. For more info contact: SMPTE, 862 Scarsdale Ave., Scarsdale, NY 10583. Tel: (914) 472-6606

NOVEMBER

12-15 **Billboard Magazine's 3rd Annual International Video Entertainment/Music Conference.** Beverly Hills Hilton, Los Angeles. For more information contact: Billboard Magazine's Conference Bureau, 9000 Sunset Blvd., Los Angeles, CA 90069. Tel: (213) 273-7040.

25-27 **Prosound '81 Professional Sound Equipment Exhibition.** West Centre Hotel, London. For more information contact: Batiste Exhibitions & Promotions, Pembroke House, Campsbourne Road, London N8. Tel: 01-340 3291.

Index of Advertisers

Coming Next Month

• In October, the British are coming! And Paul Revere will not be riding out to warn anyone! A sixteen-page editorial supplement will cover various aspects of what is happening in pro audio on the other side of the pond. This supplement appears in addition to all our regular features. Articles on Architectural Acoustics is our subject for the month. Michael Rettinger is just one of the featured authors. And there will be our new regular roster of columnists—Barry Blesser, Norman H. Crowhurst, and Leonard Feldman. All this and more coming in October in **db—The Sound Engineering Magazine.**

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Circle 36 on Reader Service Card

BARRY BLESSER

Delta Modulation: The Poor Man's Converters

• A delta modulation A/D and D/A system is just about the cheapest way to digitize audio since the parts cost can be as low as 70 cents. If such a system could be made to be high quality, digital audio would be cheap, easy, and close to "apple pie." Before showing you its defects, let's consider the way that it works. FIGURE 1 shows a simplified delta modulation system. The comparator compares the audio input with an internal approximation to it, called $a_1(t)$. We also note that the decoder has an identical circuit which generates $a_2(t)$.

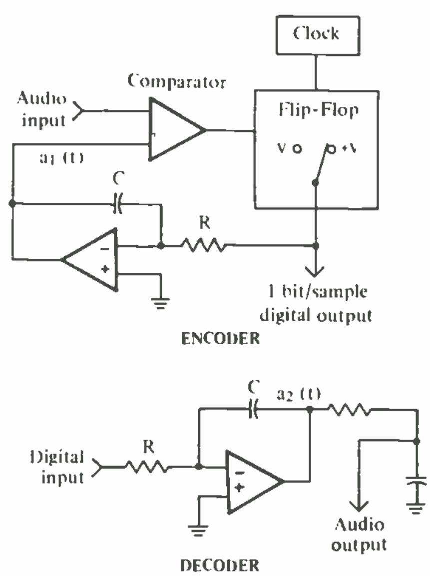
Since $a_1(t)$ and $a_2(t)$ are both created from the digital output bit from the flip-flop, they will be identical if the circuit component values are identical. For the moment, assume this to be the case. The actual conversion is performed by the 1-bit comparator which determines if the approximation $a_1(t)$ is larger or smaller than the actual audio input. If the audio input is larger, the comparator output will be an H (high = 5 volts). When the clock comes along, this information is transferred to the flip-flop's output. This is the 1-bit digital signal which represents the sign of the difference between the input and the approximation. When the difference is positive, the flip-flop's output is negative, causing the integrator's output to ramp in the positive direction (the integrator inverts). This brings the approximation closer to the input.

If the audio input is below the approximation $a_1(t)$, then the comparator output would be an L. The flip-flop output is now positive, and the integrator ramps downwards. We see that this is a kind of feedback loop in which the digital word tries to keep the error (i.e., the difference between the approximation and the input) as small as possible. Unlike analog negative feedback, the error cannot be driven to 0 since the integrator's values are quantized. Since the +V (or −V) will be applied to the integrator for a full clock cycle, the integrator will be more a fixed step downwards (or upwards). For example, with a 1 MHz clock, the integration will last for 1 μsec. In FIGURE 1, consider that V = 1 volt, C = 10 nF, and R = 10 kohms. During a 1 μsec interval, the integrator can move exactly 10 millivolts upwards or downward. We say that the change in the integrator is quantized in steps of 10 millivolts, e.g. the effective LSB size.

The error is also the "noise" in the output signal, since the two approximations $a_1(t)$ and $a_2(t)$ are, in principle, the same. For the remaining part of the discussion, we will treat these two signals equivalently. FIGURE 2 shows the response of $a_1(t)$ to a ramp-like audio input. In the region where the input is constant, the output oscillates about that value. The integrator ramps upwards for one cycle and determines that it is too high; it then ramps downwards and determines that it is too low. This kind of behavior is called hunting, or limit-cycle oscillation, and is based on the step size. It is directly analogous to the LSB quantization noise.

In the ramp part of the input, the approximation attempts to track the audio input signal, but the fastest it can go is still too slow. Even with the integrator stepping upwards every clock cycle, the error becomes larger and larger. When the audio input levels off, the approximation eventually does catch up. This kind of error is called slew limit or rate limit. This is the maximum signal limit, and is a function of the size of the encoder. Notice that the large signal limit is a rate or slope limit and not an amplitude limit. Assuming that the integrator

Figure 1. Simplified delta modulation encoder and decoder.



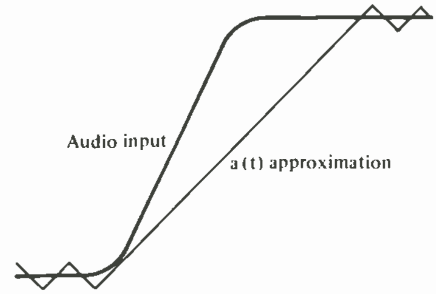


Figure 2. Example of slope overload at delta modulation decoder. Approximation is output for given inputs.

has a large voltage range, the approximation can get to any large value if one only waits long enough. This means that the maximum level sine wave at frequency f will be twice the maximum level at frequency $2f$. Each octave increase in signal frequency produces a 6 dB reduction in maximum amplitude. Of course, one could increase the step size to get larger signals, but this would also increase the LSB-type noise. A more effective alternative is to increase the sampling frequency. The $a(t)$ slope can be doubled by doubling the number of steps per second without increasing the LSB noise.

To appreciate the kind of numbers involved, let us compare a delta modulation system to a straight PCM system with the integrator step defined as an LSB. A 12-bit converter has 4000 steps between its minimum and maximum values. What sampling frequency would we need to follow a 10 kHz sine wave with the same 4000 steps? In any sine wave, the maximum slope occurs at the zero crossing, and a little mathematics will show that this slope is equivalent to 4000 steps in about 25 μsec. This translates to a delta modulation clock frequency of 160 MHz. If the comparison had been made with a 16-bit PCM converter, we would have come up with a number of about 2.5 GHz. Clearly, these numbers are extremely high and not easy to implement. Moreover, the data rate in terms of bits-per-second is much higher than the PCM equivalent.

BIT INEFFICIENCY

The bit inefficiency of delta modulation can be appreciated from the following kinds of comparisons. To double the dynamic range of a normal PCM system, we need to add one more bit to the A/D converter since this gives us a 6 dB higher signal level. At a 50 kHz sampling frequency, an increase of 1 bit is 50,000 bits-per-second increase. With delta modulation I need to *double* the clock frequency. Hence, if I were at a clock at 100 MHz, I would need to go to 200 MHz, for an increase of 100 M bits-per-second.

Actually, the dynamic range is proportional to $f^{3/2}$ rather than to f. The reason is that a doubling of the frequency also doubles the frequency over which the quantization energy is distributed. Thus,

the amount of noise in the audio band will decrease by 3 dB when the total bandwidth is doubled.

With a normal PCM system, doubling the number of bits doubles the dynamic range in dB, e.g. a 16 bits/sample PCM has 96 dB of range compared to 48 dB for an 8 bits/sample PCM. A delta modulation system gives an additive increase of 9 dB for a doubling of the bit rate.

Just as there is no free lunch, we do get something for our increase in delta modulation frequency: we increase the system's ability to handle high frequency signals *above* the audio region. A 200 MHz delta modulation system can encode small signals at 50 kHz, 200 kHz, etc. This is a useless feature for audio. The extra bit inefficiency is actually carrying information about high frequencies, but there happen to be no signals in this region. We ignore things like tape recorder bias signals and stereo subcarriers.

ADAPTIVE DELTA MODULATION

The defects just described make a classical delta modulation system impossible for audio. There is, however, an interesting variation which is usable for audio. Consider the idea of making the integrator's step size variable. For small signals we could have it be 1 millivolt, whereas for large signals we could make it 100 millivolts. This would allow the LSB to be small when the audio was small, yet it could follow a fast signal without slew limiting. This just leaves us with the question of how to control the step size.

The digital bit stream contains all the information we need. If we see a bit stream that looks like 1011111111111010, then we can conclude that the series of 1s means that the integrator can't catch up to the fast-moving audio input signal. Similarly, a series of 00000000 would indicate that the signal was moving downwards too fast. In contrast, a series of 10101010101 means that the approximation is hunting for the correct value, but cannot get to it because the step size is too large.

An adaptive algorithm might thus be expressed as the following:

A) when there are 3 successive 1s or three successive 0s, double the step size. If the 4th bit is still in the same direction, double the step size again.

B) when there are 3 alternating bits, halve the step size. If the 4th bit is still alternating, halve the step size again.

An exact analysis of such an algorithm is very complex but we can see in principle that such a system could handle both large and small signals. It is directly analogous to floating-point PCM which also changes its LSB size depending on the signal amplitude. Unlike the floating point system, the adaptive delta modulation can determine its step directly from the actual transmitted data. One does not need an additional path for the gain information.

FIGURE 3 shows the behavior. We assume that the audio signal had been large with an *a(t)* step size of 16 units. The audio becomes small and the integrator oscillates about the audio producing the sequence, 010 (circled); at this point, the adaptive algorithm says, reduce the step for the next point to 8 units. The next point is a 1 which is still alternating, e.g. 0101, so the step is reduced to 4. The process continues until the step size reaches its maximum, which is 1 unit in our example. At some time later, the audio begins to increase and there is a 111 series (also circled) which forces the next step size to increase to 2 units. Following this point, the sequence is 011011011 which is neither an increase nor a decrease. There are neither three-bits-the-same nor three-bits-alternating.

The technical implementation is slightly complex, but a digital engineer can come up with many circuits which will change the step size. An example is shown in FIGURE 4 for a system with 8 possible steps sizes each of which is twice the size of the previous one. A common implementation is to use an 8 bit D/A converter as the step size generator. The converter is not used in the usual way but as a source of 8 precision voltages. The MSB provides a

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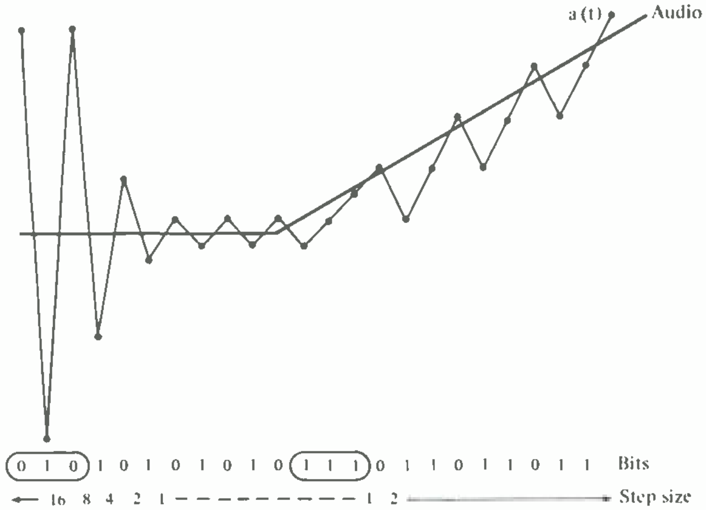
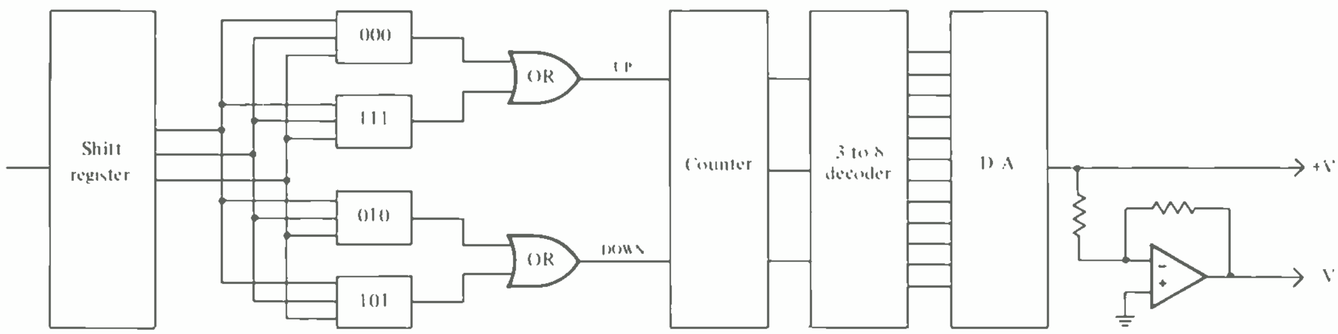


Figure 3. Delta modulation decoder outputs with adaptive step-size strategy. Audio represents the input. a(t) represents the output.

Figure 4. Block diagram of encoded algorithm for adaptive step-size.



source of 10 volts, the next bit 5 volts, the next 2.5, etc. A 3-bit counter represents which bit is to be used. The 3-bit words (e.g. 000, 001, 010, 011, etc.) feed a 3-to-8 decoder such that the 0th line is active for 000, the 1st line for 001, the 2nd line for 010, the 3rd for 011, etc. The up/down counter can thus go to a step size which is twice the size by stepping upwards one count, or it can go to a step size which is half by stepping down.

The encoded data from the delta modulator is passed into a 3-bit shift register so that three sequential bits can be examined at one time. The three bits feed digital detectors. When the "000" detector senses the correct pattern, it increments the counter; when the "010" detector finds its pattern, it decrements the counter. This circuitry must be implemented at both the encoder and decoder since the step sizes must be the same.

Systems like these have been built and the performance is quite good in comparison to a floating point PCM. Dynamic ranges of 100 dB are possible because this is only determined by the step size. Notice that with the use of 8 steps, the dynamic range improvement from step changes is 48 dB. However, when the design is complete with all of the optimizations, the hardware becomes quite complex and is no longer a bargain compared to straight PCM of comparable quality. ■

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Complexity of Sound Waves

• Acoustic waves that propagate sound are generally referred to as longitudinal, which means that movement of the air particles takes place back-and-forth along the same direction in which the wave propagates. This is to distinguish them from electromagnetic waves, in which the oscillations of the magnetic field take place along directions mutually at right angles (transverse) to the direction of propagation.

Both those terms of reference are basic. They are only fully realized when the waves are also what are called "plane waves." This means that the wave we are analyzing is at considerable distance, relative to its wavelength, from the source, so that it is no longer an expanding wave. It is the property of both kinds of wave at such distances from their respective kinds of source that enables them to radiate their respective kinds of energy so efficiently.

With electromagnetic waves, the traveling electric wave maintains the traveling magnetic wave, and vice versa, each mutually at right angles to the direction in which both of them are traveling, or propagating. With acoustic waves, alternations of pressure above and below mean atmospheric pressure are accompanied by momentary particle movements, forward and backward, along the direction of propagation. The pressure changes and particle movements are mutually self-sustaining.

This means that momentary particle movement forward conveys the increase of pressure forward, while momentary particle movement backward conveys the reduction of pressure forward. At the same time, pressure higher than mean atmospheric causes forward particle movement, while pressure lower than the mean causes backward particle movement.

The whole wavefront is maintained because the waves we are considering are plane waves. This means that all points in the same phase of propagation are in a plane at right angles to the direction of propagation. If adjacent points in the plane all have the same instantaneous pressure, there will be no lateral particle movement, only movement to or from other planes, where the pressure is higher or lower; particle movement occurs because of pressure difference.

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The British Are Coming!

In our October issue, we will be featuring a special sixteen-page supplement on the current state of British-made audio. Among the articles included will be "The British Audio Industry Today," "The British Sound—Truth or Legend," "Britain's Recording Equipment and Its Reputation Abroad," "Digital Recording in Britain," "Creating a Sound Studio in Residential London," and (of course) many more. This special and exclusive-to-**db** supplement has been produced by the British Trade Council, and we are sure you will find it of extreme interest. See also our *Coming Next Month* feature on page 5 for all the other articles that will be in October.

Pressure is a function of the elasticity, or compressibility, of the medium through which the sound travels, generally air. But if this were its only property, acoustic waves would be no more complex than the pressure changes in a bicycle pump, and waves would not propagate. The other property of the medium is its density, or inertia, of the particles that make it up.

Because of elasticity, pressure differences cause particle movements from the higher to the lower pressure. Because of inertia, particles in movement convey changes in pressure from point to point. Pressure, as from a bicycle pump, causes air particles to flow from higher to lower pressure. And air particles in movement, such as in a wind, cause an increase in pressure when some obstruction slows them down.

Those effects can be observed separately, but acoustic waves occur when they both happen together, each mutually supporting, or causing, the other. And this happens at a characteristic velocity, which in air is a little over 1000 feet per second.

The only way to generate a plane wave would be by using some form of plane, theoretically infinitely large, and vibrating the whole surface back and forth. That is not very practical. Sound waves must come from sources of finite size, often much smaller than the wavelength of the sounds the source propagates. So we have to consider kinds of source other than this theoretical infinite vibrating plane.

In acoustic theory, the next kind of source theorized is the so-called spherical source, which can be visualized as a small balloon, in which the air is rapidly compressed and rarefied. This causes the surface of the balloon to rapidly become larger and the smaller.

With the plane wave, pressure changes and particle movements occur in phase. With a spherical wave near to its source, which means where the distance from the theoretical point source (which a small balloon only approximates) must be small compared to the wavelength of the sound being radiated, particle movement must be much greater than pressure changes, just to get the air necessary to cause those pressure changes, "in and out" of the point source. This increased particle movement, close to the source, is not in phase with the pressure changes associated with it, but gets to be almost in quadrature. Only the energy component is in phase.

So far we are thinking of a spherical, or point source, acting in an infinite medium. As you move out into the medium, a section of the wave more closely approximates a plane wave, with the result that pressure and velocity get more closely into phase. But now, if you obstruct the wave, within the range that may be considered as "close to a point or spherical source," the higher velocities

encountered there will cause pressure changes due to the presence of the obstruction (just as a fence or wall creates pressures when a wind strikes it). And like a fence or wall, some of the air is diverted over the wall, instead of building up pressure forever.

The final basic kind of acoustical source is a dipole, or a pair of point or spherical sources. This can have variations, as we shall see. You can think of a dipole as a pair of small balloons, one of which expands as the other contracts, and vice versa. Another way to think of it is as a tiny piston that vibrates back and forth in open air, creating pressure on one side and rarefaction on the other at one moment, reversing it the next.

Now perhaps we can consider these basics in relation to some types of transducer, with which audio people associate them. A lot of small loudspeaker units, mounted in a plane and connected in phase, will radiate a close approximation to a plane wave, at least directly "in front" of it. But around the edges, because the radiating "plane" comes to an end, the condition we described earlier as belonging to a plane wave will no longer apply. Particle movement will "escape" from the edges of the pseudo plane wave into the surrounding medium which is not being moved in the same way. But the central part of the plane wave will maintain its direction better. This is the kind of system attributed to Bose.

If, instead of a plane, the units are arranged in a line, the end units will have the same kind of velocity "spill out," but the middle units will spill out

at the sides of the line, not lengthwise. And as all the middle units will have the same sideways spill out, the wave radiated will approximate cylindrical, with the axis of the cylinder coinciding with the line of units.

Now think about what happens with a loudspeaker unit in which both back and front are open like a small vibrating plane: not a small piston, which has small dimensions compared to any of the wavelengths it radiates, and not big enough to radiate plane waves from back and front, out of phase. What happens here is that air rushes back and forth around the edges in particle movement that is in quadrature with the pressure that accompanies it (a sort of squashing action, like the water in some washing-machine agitators).

This becomes a type of transverse wave. The sound wave moves away from the unit, but the particle movement remains edge-on to the direction of propagation, and also in quadrature with it. Close to the edge, particle movement is considerable, but it falls off rapidly with distance.

The energy component of the radiated wave can best be thought of when the distance from the source is very large compared to wavelength. Although it now approximates a plane wave, it is really a small part of an expanding sphere. So when the diameter of the sphere doubles, its surface area quadruples, and the energy in a small piece of it, say one square foot, will be reduced to one fourth. This is what is meant by the "inverse square law." Energy intensity is inversely proportional to the square of the distance from the source.

Such waves are always close to longitudinal, and pressure and velocity are always close to in-phase. Quadrature components of velocity are always much bigger than these in-phase components, where the quadrature components occur. And because they reduce much more rapidly than according to the inverse square law (involving inverse cube terms), they disappear as distance from source increases, leaving only the longitudinal component.

In a position edge-on to a loudspeaker unit with open edges, and at some distance away, the transverse component has virtually disappeared and there is very little longitudinal component anyway. So theoretically, there will be zero sound in this direction, at any distance from the source. The fact that you can hear it is due to reflected sound, either from obstructions close to the unit, from other parts of the room, or both.

This can be verified by careful measurements in an anechoic chamber. But now let us turn to think about how we hear. If you think about the outer and middle ear structure, you will realize that the human ear is an almost perfect pressure transducer. It knows nothing of sound particle velocity, directly. A compression travels the short distance from the side of your face to the tympanic diaphragm, which it depresses in order to compress the air in the middle ear (long-term averaged by the Eustachian tube) and the movement of the diaphragm thus caused is transformed by the 3-bone structure for transmission to the inner ear, and nervous transduction.

Sound from directly in front, behind, overhead, or any direction such that it reaches both ears absolutely simultaneously, will send messages to the brain via both ears simultaneously. Sound from one side, to go to another extreme case, will produce two or maybe even three effects that the brain can observe. In this position, the particle movement is such that the head constitutes a more definite obstruction than it does in the other set of positions.

In the equidistant set of positions, the sound wave flows over both ears, so that each of them can "sample" the pressure fluctuations as it goes by. When sound comes from one side, the particle velocity comes up against the near side of your head and the obstruction effect translates that into sound pressure, more than would be encountered in the other directions of approach. But on the far side of your head, a much reduced pressure is generated.

There is also a difference between the times at which the waves arrive at the two ears. When it comes from one side, it arrives at the nearer ear sooner. Other things happen to the sound wave, at each ear, dependent on the shape of your face and especially those external appendages that we call "ears," but which contribute nothing directly to the transduction. However, they do modify the sound wave due to its particle movement, and thus provide more clues as to direction than you could get if your head merely had plain holes in it!

So far we have talked about radiating sound, and about how we hear the simplest forms of arriving sound. But when we get into stereo, and stereo reproduction, there are more interactions that happen. The system we use to radiate the "stereo" can now produce more complex patterns of particle movement that can be used to create illusions for you, using these properties of hearing in more complex ways. That is what we want to get into next. ■

LEN FELDMAN

db Sound With Images

Stereo TV—What's the Holdup?

• Japanese TV broadcasters have been transmitting multi-channel audio along with their video signals for the better part of three years now. Note that I called their audio transmission multi-channel, rather than stereo. That's because the transmission system is used not only to transmit stereo audio (which at the moment is limited almost entirely to videocasts of concerts and other musical events), but also as a means of transmitting bilingual audio information. In fact, if you were to pick up the equivalent of a TV Guide Magazine in Japan, you'd probably find more of these bilingual transmissions than actual stereophonic audio transmissions on a typical evening in Tokyo or Osaka. But that is no surprise, since there is not a great deal of video fare (either in Japan *or* here) that lends itself to stereo audio. Back in the early 1960s, our own Federal Communications Commission, when asked to consider the possibility of stereo audio for TV, sagely decreed that no one would want to watch a tiny video screen and hear sounds coming from "way out to the left or right." That was before the days of large screen projection TV, of course. I know a great many people, though, who even with their "small screen" TV sets enjoyed simulcasts of concerts in which the video was broadcast over Public Television, and the audio transmitted over a local stereo FM station. Evidently, the argument that people won't enjoy watching a small picture while listening to a big sound doesn't hold water.

Well, rather than wait for us, the Japanese, in their by-now familiar way, went right ahead and developed a perfectly workable system for two-channel audio transmission and managed to effectively squeeze it compatibly into the same NTSC video format that we use in this country. (They adopted the NTSC system at a time when they didn't know better and were still taking technological cues from us, I think.) Interestingly, in the case of stereo FM, the Japanese also took their cue from us and

ended up with the same AM-suppressed carrier system that we use. I suspect they learned from this experience and went ahead on their own when it came to deciding upon a multi-channel audio system for TV.

Having heard the system in Japan on several occasions, I can tell you that it works extremely well. And, of course, it generated a whole new group of products, such as stereo TV tuner-only units (which can be added to existing color TV sets), add-on amplifier/speaker units and, last but not least, new top quality color TV sets which have the multi-channel decoding and playback facilities built right in along with left- and right-channel speakers. It is a real joy to be able to switch from Japanese-dubbed dialogue, heard with an American made film, to the original English sound track. And, of course, this added capability means that there is no need for sub-titles, which I often find so difficult to read, especially on a TV screen, when contrasts obliterate the titles.

The Japanese multi-channel TV audio system employs an all-FM system. That is, both main-channel and sub-channel are frequency modulated. Sum-and-difference techniques are used when stereo is being broadcast, while for bilingual applications, the main language is transmitted over the main channel while the alternate language is transmitted via the sub-channel. Simple—and it works.

MEANWHILE, BACK IN THE GOOD OLD U.S.A.

In this country, the FCC has had second thoughts about the whole question of multi-channel audio on TV, and so now, the Electronic Industries Association (EIA, a trade association, one of whose groups embraces consumer electronics manufacturers) is hard at work, with several committees attempting to develop a report for the FCC which will enable it to promulgate transmission standards for multi-channel audio on TV.

By now, you may be wondering why such committee work is needed if the Japanese system seems to work so well in Japan. After all, audio and video electromagnetic radiation works about the same in Asiatic longitudes as it does in the Western hemisphere. The reason for the deliberations is that more than one system of multi-channel transmission is being evaluated by the committees. In fact, there are three. The system now on-the-air in Japan is being sponsored as an official entry in this country by the EIAJ (Electronic Industries Association of Japan) a trade association that is the equivalent of our own EIA. But two other systems also proposed are those developed by Zenith Radio Corporation and Telesonics Corporation. To be sure, having competitive systems is certainly in our American tradition, and if one of the systems is better than the others, the FCC should certainly arrive at that same conclusion and make it the official standard for the U.S., regardless of what is done in other countries.

Both of the competing systems proposed by U.S. companies utilize an AM subcarrier, for reasons we won't go into just now. But as anyone who is experienced with stereo FM knows, any multi-channel system using sum-and-difference techniques (L+R on the main channel; L–R on the sub-channel) and an AM subcarrier is going to end up being noisier (at weak signal locations,

out in the suburbs) than a system which uses an FM subcarrier, all other things being equal.

So now, many people have become concerned about reduced audio coverage. In all fairness, it should be pointed out that even with an all-FM system, some increase in noise would be inevitable (you hardly ever get something for nothing, and cramming two channels where only one used to be has got to mean *some* tradeoffs). With that in mind, it was proposed that perhaps the answer was to do a bit of electronic noise reduction—build the noise reduction right in, as a standard, right from the beginning. So, the call went out for various noise reduction folks to come forward as proponents. And before long, there were three of these: Dolby, dbx, and CBS, with a variation on their new CX noise reduction companding system. dbx had to modify their familiar linear companding system because one of the most important criteria for any system to be approved by the FCC would be that it had to be "compatible." In other words, owners of present TV sets must be able to receive their TV audio information without having to pay for anything extra—and that audio should sound about like it always did. Well, if you've ever heard un-decoded dbx-encoded material you know that 2:1 compression is not very pleasant to listen to *unless* you decode it (1:2). So, dbx dutifully modified their system to make it compatible. The CBS CX system was already compatible (that's one of its claims to fame—you don't have to buy a decoder to listen to CBS-CX discs, but you won't get the noise reduction benefits unless you do). As for Dolby, all they had to do was decide whether they wanted to offer Dolby B (with its 10 dB of noise reduction at high audio frequencies) or their newer Dolby C (with as much as 20 dB of noise reduction). They chose Dolby C.

So, here's where matters stand. We now have three basic transmission systems proposed for multi-channel TV and each of those systems may be augmented or accompanied by one of three proposed noise reduction systems. You could have the EIAJ system with the Dolby noise reduction system, or perhaps the Zenith system with the CBS CX companding system or... well, I'm sure you get the idea.

It should be pretty clear from all of this that it's going to be some time before the FCC gives the go-ahead for a particular system. It is even conceivable that they may not choose one of these nine permutations at all, but might say to the industry (broadcasters and consumer hardware makers alike) that they are free to experiment with any combination they choose. This is the so-called "Let The Marketplace Decide" approach which has been mentioned from time to time in connection with the still-awaited stereo AM decision and the long-postponed (and almost forgotten) quadraphonic question.

I don't know whom to pity more while we wait—American TV broadcasters, or Japanese hardware manufacturers. The broadcasters would like nothing more than to have the extra capability of stereo and/or bilingual audio. As for the Japanese manufacturers, right now they have to make two basic kinds of models: twin channel versions of video products for their own domestic consumption, and (from their point of view) somewhat archaic mono models for export to the rest of the world and to the U.S. in particular. And remember, everything we've talked about applies to VCRs as well as to basic TV sets. I must confess that I was somewhat amused a few months ago when one maker of VCRs introduced into the U.S. market a video cassette recorder that "actually had stereo recording and playback capability." That must have seemed like quite an advance to the uninitiated here. To those of us who were aware of what goes on in Japan, it was nothing more than a clever way to save money by not having to have two basic types of VCRs. ■

Broadcast Audio

PIFO & PAPALLACTO, August 1981. On the off-chance that some of our readers may not instantly recognize our dateline cities, the former is a small village some twenty miles outside of Quito, Ecuador, while the latter is a still-smaller village some twenty miles beyond Pifo. Papallacto used to be the end of the road: if you wanted to penetrate further into the Ecuadorian interior, you could rent a mule here for your continuing journey.

Times have changed. The road now extends just a little further into the jungle, and there's a new hydro-electric plant just outside of town. The plant supplies electricity for a 500 kW transmitter at Pifo. That's right, 500 kW. You might say that the folks in Pifo have really taken the NRBA's "We're for Radio" slogan a little too seriously. But not quite: Pifo happens to be the transmitter site for the legendary HCJB Radio. Here, "La Voz de Los Andes" broadcasts in some 14 languages to listeners as far away as Siberia and Japan. It's quite a story, and you'll read more about it in our March, 1982 International Audio issue.

As luck would have it, this month's broadcast audio editorial came due just as we were making our way through the Andes from Pifo to Papallacto. (Actually, it was due before you left New York—Publ.) In this little corner of the world, it's difficult to comprehend just how close we are to New York. In space, we're about as far from home as if we had gone off to Los Angeles. Our closeness in time is quite another matter. The dam and the transmitter it serves are twentieth century—all else here is not. It's a difficult place from which to write a 1981-style editorial. The pressures of deadline-meeting, and other clock-related matters, don't seem very important—maybe it's the altitude.

As we left home, there were rumors that our fast-moving FCC was about to leap into action on the subject of FM quad broadcasting. Now that most of us have forgotten what that is, presumably it's safe for them to make a decision. Since they've now had some ten years to think it over, let's hope their actions will be a little more carefully thought out than the recent AM stereo comedy of errors.

While we're waiting for the learned Commissioners to lead us down the path to higher technology (or failing that, at least to stop blocking traffic), it's refreshing to note the increasing interest by everyone else in higher quality broadcast audio. The signal path within the broadcast studio used to be one of the weaker lines in the audio chain. But lately, the line is getting stronger all the time. As many of this month's feature stories illustrate, prominent manufacturers of recording studio consoles have responded to the demand with specially-designed boards to meet requirements of the quality-conscious broadcaster.

It's no longer enough to be the loudest station on the dial, and broadcasters are discovering that a sizeable segment of the audience will tune in to quality when and if it's available. It has been discovered that most listeners can manage to adjust the volume control to get the loudness they desire. For those stations that seek to spare the listener this inconvenience, by broadcasting loudness and little else, some of those listeners are responding by adjusting the station selector knob instead.

Of course, the loudness race has by no means been called off entirely, nor is this likely to happen in the immediate—or even distant—future. But the days when broadcast signal processing required only a brute-force compressor are fast drawing to a close. Broadcast audio is now getting the kind of serious attention it deserves. Here at db (that is, at our Papallacto branch office), we're glad to do our little bit by presenting this broadcast audio issue.

GEORGE BERGER and ROBERT L. DEITSCH

Broadcast Audio— The Increased Need for Audio Processing

Here, authors Berger and Deitsch provide us with a behind-the-scenes look at the problems (and the solutions) faced by engineers in the broadcast field.

THIS ARTICLE will deal with the basic concepts which underlie the design of facilities for the ABC radio broadcast studios in New York. Since 1975, when a slow but major reconstruction was begun, we have argued with our friends about what we believe are the fundamental and philosophically different aspects of broadcasting which distinguish it from the rest of the audio field. There are to this day several misconceptions about what broadcast audio is and ought to be, which still disturb audio professionals and amateurs alike.

Is there any reason why that which is broadcast should be identical to the original source? Well, there are two parts necessary to the answer. The first, and we grant some nit-picking, is that none of the records and tapes which are our primary sources are unprocessed, and secondly, shouldn't the broadcaster be granted latitude in the same sense in which recording engineers use processing to compensate for the limitations of their media or to create a uniqueness of sound for a group? We suggest that to answer no is intellectually dishonest, and thus we have ceased to feel any necessity to defend our position. We must make it understood, but the proof comes from that understanding *a priori*.

Our somewhat dogmatic philosophy about our studios is that an on-air control room should be as specifically tailored to the format as possible, and that a production facility must be so general and versatile that it never limits the creativity of either the producer or technician. Technical equipment should be of the highest possible quality. All of our installations since 1975 have been centered around custom-designed mixing desks from Rupert Neve, all reel-to-reel tape decks have been Studer A-80 mastering recorders.

It would be unfair for the reader familiar with the two ABC-owned radio stations in New York to be saying, "What a waste of good facilities: by the time I hear the product it sounds very different." The AM station is WABC which, over the years, has been the premier clear channel station in the U.S. ABC's FM station in New York is WPLJ, which is the top-rated album rocker in New York. The two stations provide some excellent examples of the use of audio processing within our medium.

EASE VERSUS QUALITY

One of the problems broadcasters have faced since the early sixties has been the conflict between the desirability of using tape cartridges for ease of production and the degradation of quality, especially in stereo, which seemed to be inherent in using them. The two major problems were phasing and degradation of signal-to-noise ratio. As early as 1973 we were experimenting with a matrix encoding system to minimize phase instability. In 1975 we decided to use dbx® noise reduction as well.

There has been much debate over the relative merits of dbx and Dolby noise reduction. Because so much of the source

George Berger is chief engineer WPLJ/NY & project manager AM stereo ABC radio division. Robert L. Deitsch is the senior design engineer ABC radio, NY.

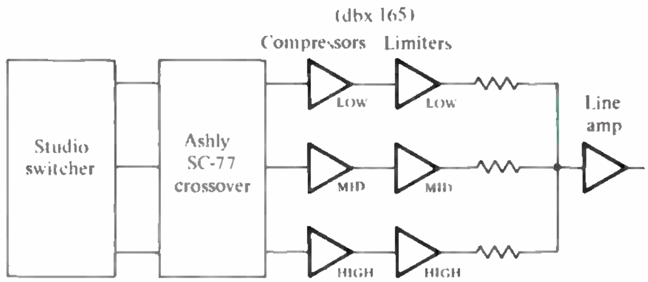


Figure 1. WPLJ's "better mousetrap" consists of an Ashly SC-77 Crossover and six dbx model 165 "Over Easy" Compressor/Limiters. (A) Block diagram; (B) rack-mounted equipment.

material we use comes to us already compressed, and also because dbx offers a far greater amount of noise reduction to precede the additional compression which we use, dbx was to us a far preferable choice. Both of our stations now use this combined processing system. This type of processing should be transparent; the listener, either in the studio or at home, should not be able to detect its presence.

The type of processing which causes controversy is compression, limiting, and clipping, as well as any equalization or reverb which may be added. It should be realized that the "sound" of a radio station is an important programming consideration. Psychoacoustic analysis is one of the elements which enters into such decisions, but the broadcaster must also consider competitive loudness and dynamic range.

Many listeners are in environments with high ambient noise levels, such as automobiles; a station can ignore this part of its potential audience only at its peril. On the other hand, if listeners find the signal unenjoyable because too much processing is added to "fix" this type of problem, audience may also be lost. Consequently, the final decisions, within the limits imposed on all broadcasters by the FCC rules and regulations, must be made among what are still a scientifically ill-defined set of possibilities. Having made those tough choices, the broadcaster must enter into the area of receiver technology in order to determine how to best accomplish those aims.

It is reasonable to look at the FM broadcast chain, including the home receiver, as offering the potential of true reproduction. But it is by no means true that all FM receivers can reproduce what is in the broadcast signal with true high fidelity. Thus, even with FM, the broadcaster must attempt the clearest possible definition of the audience. With AM, the waters become much more murky. AM transmission is for all practical purposes capable of equal fidelity! It is, of course, subject to atmospheric and electrical interference, but the broadcast signal is high fidelity. THD and IMD figures of less than one percent are achievable. Signal to noise ratio approaches 60 dB. Today, most AM receivers begin to roll off frequency response at 1 kHz and are down more than 6 dB by 5 kHz.

In defense of receiver manufacturers (but certainly not to agree with them), it is, of course, more expensive to build wide response high fidelity AM receivers. Most firms claim that there is no mass market for such equipment. We as broadcasters counter that this is a matter of consumer education; there was no immediate large-scale rush to FM equipment either.

So, we the broadcasters are left attempting to compensate for the varying quality of receivers, both AM and FM, that are in the hands of the public. Both of our New York stations play "popular" music. The problem faced by the technical staff to meet the desires of the WABC programming department is to provides a sound which makes WABC-AM sound as good on the radio as an FM station. This means that we must compensate for the "poorer" AM circuitry in AM/FM combination radios and receivers.

Through experimentation, we have found that no single equalization curve can accomplish our purpose. The single most obvious objection to a simple equalization attempt comes from the announcers and disk jockeys. They feel that they don't sound like themselves. They are correct. We have found that a multiband compressor/limiter system has been the answer.

Many of the ideas for use of equipment start at our visits to Audio Engineering Society conventions. For example, we were extremely impressed with the "Over Easy®" compression curve used in some dbx equipment. Many years of experience have taught us that the use of any single band compression device does not yield the desired audio density without destroying the clarity of tone at the lower and higher ends of the frequency spectrum. Such single-banded attempts typically yield muddy drums and cymbals.

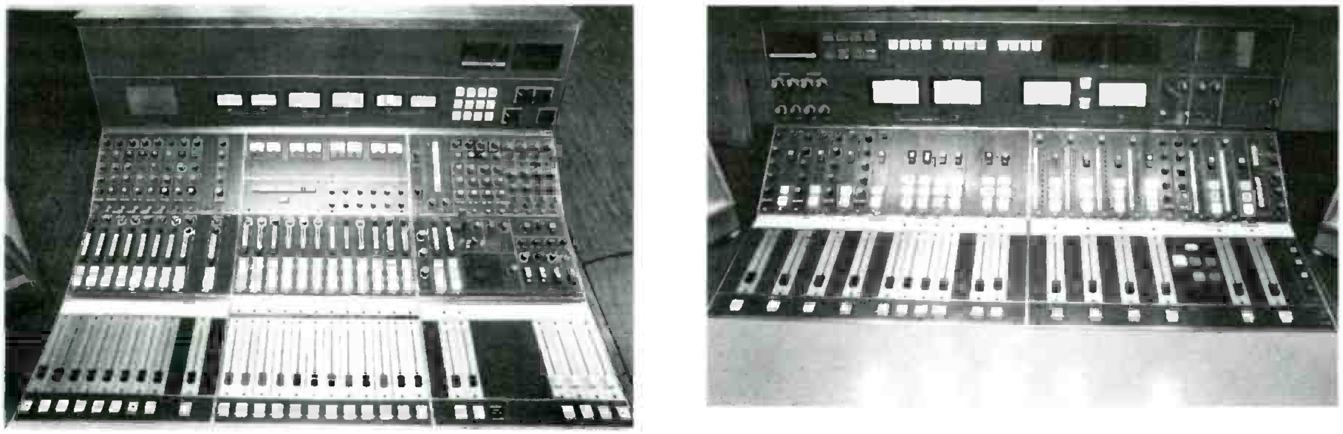
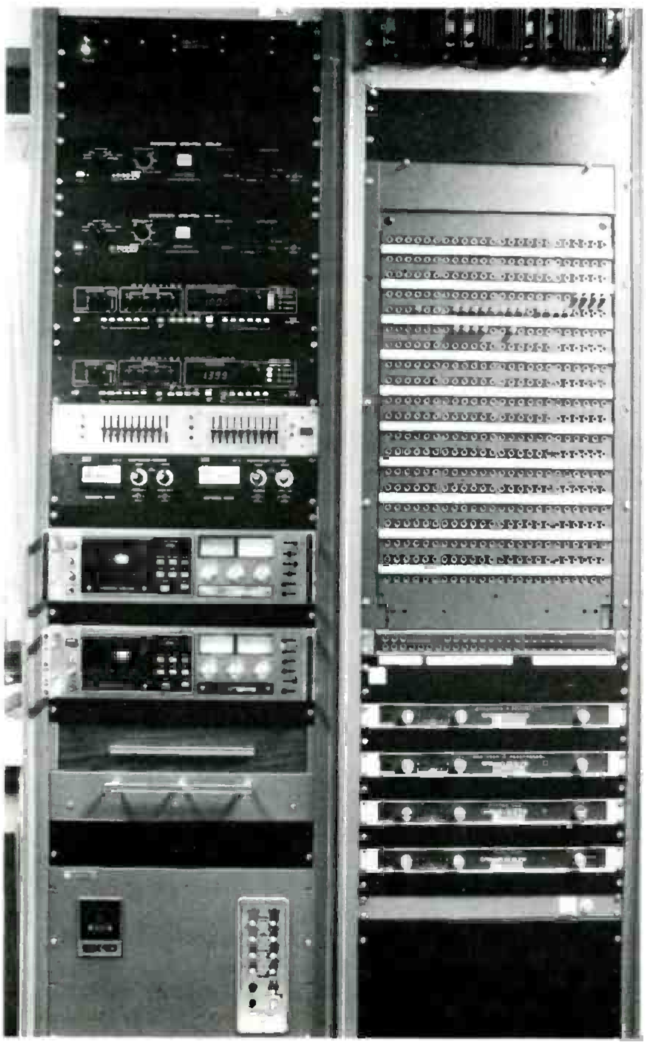


Figure 2. Custom-designed Neve boards. (A) Studio 8C; (B) Studio 8X.

Figure 3. The equipment rack in Studio 8C.



For many years, WPLJ-FM has used Dorrough discriminate audio processors. But all of our attempts to use them and other readily available tri-band processors on AM had yielded, what were to us, unsatisfactory results. We knew that it was relatively easy to make music sound good, but found that the quality of voices was changed too much. Finally, we bit the bullet and decided to attempt to build a better mousetrap out of commercially available components.

BUILDING THE PROVERBIAL MOUSETRAP

We found that an Ashly crossover with settable depth and 12 dB/octave crossover curves was best. Steeper curves didn't sound as good to us. We wanted to assemble a tri-band processor with compressor and limiter in each band to use in our air chain at the studio, plus a single processor for final gain compensation at our transmitter. A separate compressor and limiter is used in each frequency band to achieve a uniformly loud sound without pumping.

The dbx 165 offers a number of advantages beyond the fact that we liked its sound. It can be coupled for stereo, which probably is coming to AM. It turns out to be extremely important that it has user access to the limiter sidechain. It also offers externally accessible automatic or variable attack and release times, infinitely variable compression ratio, and adjustable level threshold.

The crossover is adjusted to split the frequency spectrum optimally to allow the compressor/limiter pairs to be set for thumping lows, clear midrange, and crisp highs. For most formats, we feel that two crossover points are sufficient. For disco, one might want to add an extra band for very low frequencies. We feel that different formats will be helped by adjustment of both crossover points and depth. Delay and equalization are inserted into the side chain to eliminate phase-related interactions of the recombined audio. Small amounts of delay added to the side chain signal can achieve some remarkable results. Equalization can, for example, aid in de-essing; this is important before clipping when attempting to maximize loudness in the baseband. A truly flat AM signal yields a muddy sounding AM signal because of receiver design. We have achieved a psychoacoustically pleasing sound which simulates wideband reception on most AM receivers.

The program director of WPLJ wants a compressed sound. This is achieved using a Dorrough 310C discriminate audio processor with modified recovery times, Moseley limiters, and composite clipping. When composite clipping is used, the broadcaster must adjust it so that he can pass an FCC proof-of-performance with it in the circuit as it is for everyday transmission. WPLJ is the loudest FM station in New York.

We have designed our technical facilities to allow for rapid modification of air sound. We are proud of the quality of our plant and of our ability to provide the sound required by our programmers. ■

Portrait of a Quiet Family

Meet the members of our Studer console family. The compact, very portable 169. The expanded 269. And the full size 369 studio production console. A respected family with three common traits...

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Studer consoles are designed for quiet operation, the utmost signal clarity, and precise tailoring of the sound. The inputs feature 3 section EQ, the overload margins are generous throughout, and each master module has a built-in limiter. As for S/N, well, it's no surprise that the 169 has been used for many digital classical recordings.

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If you're ready to move up to a specialized mixer, you're ready for Ramsa.

The Sound Reinforcement Specialist: Ramsa WR-8716

When your sound says you're professional but your mixer doesn't. When you're wasting your subtlety and style on "make-do" boards. When you're creating compromises instead of clear-cut distinctions. Then you're ready for Ramsa — the new mixers that are specialized so you won't have to compromise.

The WR-8716 is a fully modular sound reinforcement console with 16 input modules, 4 group modules, and 2 masters. It features 16 input pre-fader solo buttons, 4 group modules with pre-fader insertion patch points, and lockable post-fader solo buttons. There are 6 illuminated VU meters with peak LED's for easy outdoor reading and a separate stereo variable frequency EQ for monitor sends. Pan pot controls allow panning to the left or right masters while level controls permit 16 x 6 board operation. The left and right direct channel assign function lets you bypass the group modules for individual sources. Portable operation is a snap with easy access connectors.

And the WR-8716 features plastic conductive faders for greater reliability and smooth, low-noise operation; external power supply for light weight, and switchable 48V DC phantom power for condenser mics.

RAMSA

The Recording Specialist: Ramsa WR-8816

The WR-8816 recording console includes the same modular construction, input modules, power supplies, and faders as the WR-8716 plus many important recording advantages. Like direct outputs for 4, 8, or 16 track recording and peak-reading LED meters that let you monitor any 4 out of 24 signals with clear, quick response.

You'll command a variable frequency EQ section with 3 frequency settings for the high and low frequencies plus continuously variable midrange. Stereo echo send replaces the separate mono controls you'll find on competitive boards. And you get two independent stereo monitor controls—one for musician's headphones, one for control room monitors—a special feature for any mixer in this class. And there are other important features like low noise electronically balanced mic inputs with new high-speed IC's, 16 switchable post-fader solo controls and XLR-type mic connectors.

Ramsa offers a full line of specialty mixers including the more compact WR-8210 recording mixer and WR-130 sound reinforcement mixer. So don't hold down your professional sound, call (201) 348-7470, because you're ready for Ramsa.

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THOMAS MINTNER

Broadcast Audio Production Consoles

The Studer series 900 console is one way to integrate the various requirements of multitrack music recording and advanced broadcast production into one console system.

As the emphasis on complex, higher quality production for broadcast audio increases, choices involving multitrack recorder/console systems become more difficult. In order to illustrate some of the problems which designers and users must resolve, a brief review of console evolution and a description of a new console system are given.

To better explain broadcast production console requirements, some further definition may be in order. For purposes of this article, we will define broadcast production as: support audio production for radio and television, including station spot and advertising production, multitrack and simple stereo program recording for later broadcast or post-production and also such related audio tasks as video sweetening, film work, etc. Broadcast production should not be confused with the recent term "production equipment," coined by manufacturers to refer to equipment formerly called "semi-professional," although certainly some equipment of this type is in use today for broadcast production. We will also exclude from discussion the "air console" used for normal broadcast, usually of a considerably different (often simpler) pattern than the production console.

In radio's earlier days, the production console was often the station's previous air console, now relegated to the back room with a similar collection of recorders. This arrangement was adequate for bread-and-butter spot production and similar activities.

In the television area, sophisticated audio production was formerly quite rare, being limited mainly to networks or network owned-and-operated affiliates where a large audio console for production activity was usually custom designed.

As recording studio activity flourished in the late 1960s, equipment manufacturers brought multitrack facilities in one form or another to more and more users. Eventually, there was a demand for better audio in broadcast. The industry responded quickly; today in scanning the pages of broadcast journals, it is easy to see that audio signal processing and control is a subject easily holding its own among the other major broadcast concerns such as ENG and distribution by satellite. Indeed, the new distribution technologies, including analog and digital audio channels of very high quality, have effectively removed the old excuses for letting broadcast audio quality remain at lower levels, and thus serve as a further impetus for more sophisticated audio production.

While large broadcast operations may be able to afford custom-designed hybrid consoles with multitrack capabilities combined with necessary broadcast features, many other users have been forced to outfit themselves with small general purpose recording consoles or with the one basic architecture of the studio music recording console where little thought is given to combination use. Some manufacturers make both types of consoles, others only one. The first type is the conventional production or small recording console, with one to four main mixing buses, for use with 2- or 4-track recorders. This type of console is often supplied to broadcasters as-is, or modified with a number of useful items such as stereo line level input modules, multi-input selector switches, broadcast type cue features and fader start switches or other remote control facilities for tape recorders.

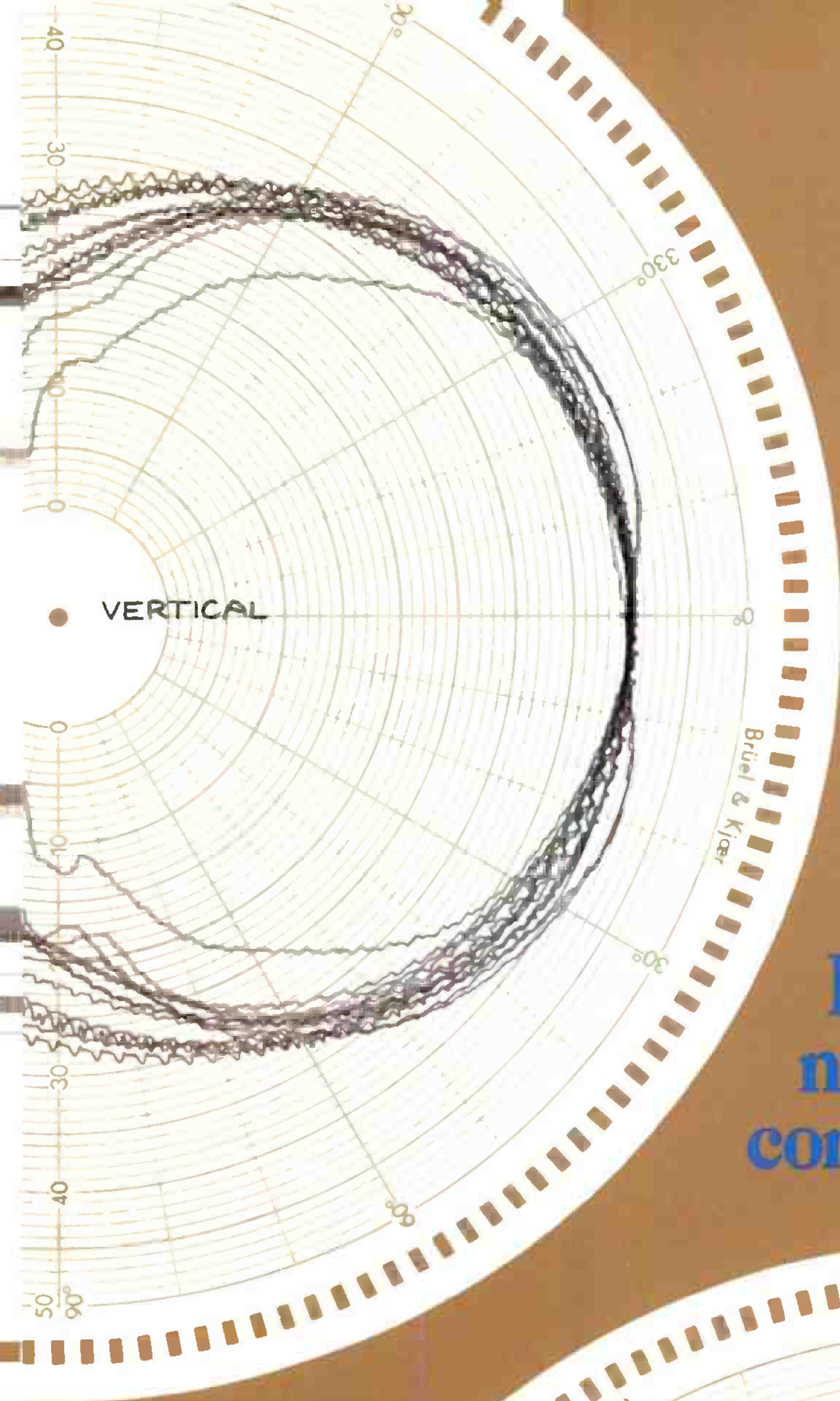
Generally, the basic philosophy is that inputs (typically 12 to 16 in this size console) will be simply mixed live to stereo (or possibly to a 2- or 4-track recorder), with monitoring in mono, or stereo derived from the main mixing buses or tape output.

At the time when the recording industry was first expanded to 8 and 16 track operation, this type of console could still be used in studios, with the relatively minor additions of an independent multitrack metering bridge and some type of monitor arrangement for the multitrack recorder. This might be a simple external 8-by-2 or 16-by-2 matrix mixer which could then be selected on the monitor panel. This was early multitrack production with the following characteristics: the number of mixing buses was much smaller than the number of recorder tracks, a simple multitrack monitor allowed for multitrack listening during tracking, there were no direct outputs from input channels, and much patching was required.

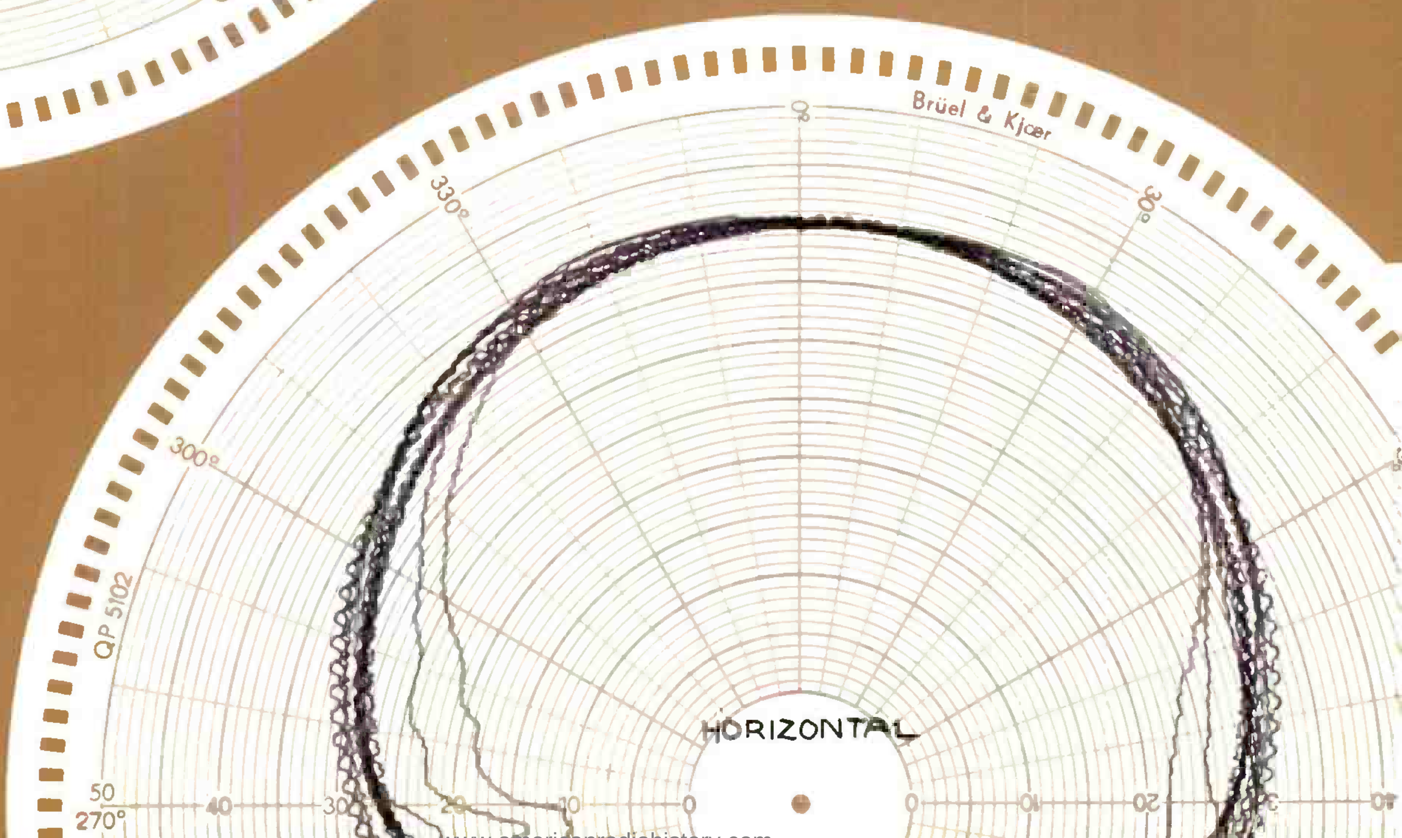
Gradually, the console described above pretty much disappeared, as the increasing demands of music recording led to the evolutionary development of a "Mark II" console, which in general had the following characteristics: the number of mixing buses is equal or near to the number of recorder tracks, there is full metering for the buses and multitrack, internally switchable, and there is a well-developed multitrack monitor section, either in-line or as separate side monitor

Thomas Mintner is the manager of Broadcast Products with Studer Revox America, Inc.

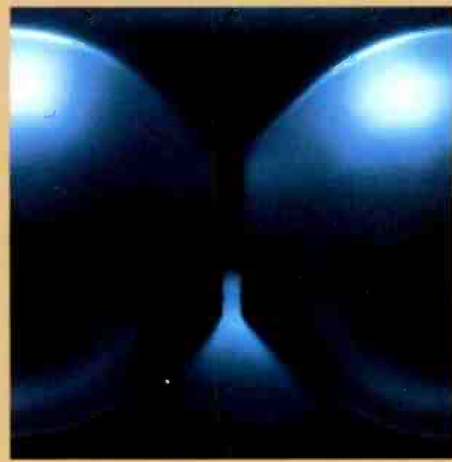
Circle 22 on Reader Service Card ⟶



Before you invest in new studio monitors, consider all the angles.



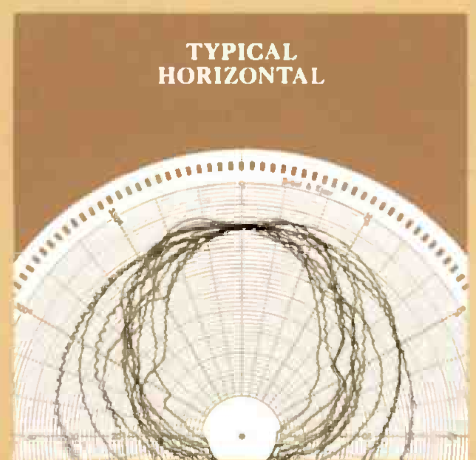
Introducing the JBL Bi-Radial Studio Monitors.

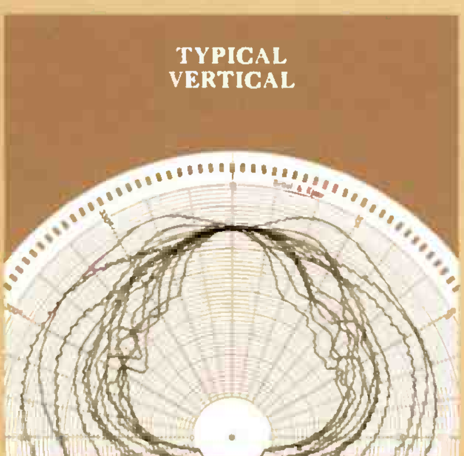


No one has to tell you how important flat frequency response is in a studio monitor. But if you judge a monitor's performance by its on-axis response curve, you're only getting part of the story.

Most conventional monitors tend to narrow their dispersion as frequency increases. So while their on-axis response may be flat, their off-axis response can roll off dramatically, literally locking you into the on-axis "sweet spot." Even worse, drastic changes in the horn's directivity contribute significantly to horn colorations.

Polar response of a typical two-way coaxial studio monitor:





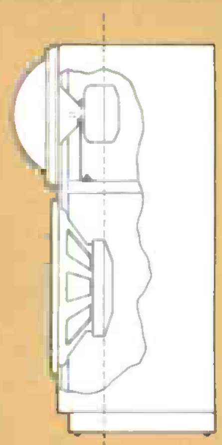
At JBL, we've been investigating the relationship between on and off axis frequency response for several years. The result is a new generation of studio monitors that provide flat response over an exceptionally wide range of horizontal and vertical angles. The sweet spot and its traditional restrictions are essentially eliminated.

Polar response of a 4430 studio monitor:

The Bi-Radial Horn

The key to this improved performance lies in the unique geometry of the monitors' Bi-Radial horn.[1] Developed with the aid of the latest computer design and analysis techniques, the horn provides constant coverage from its crossover point of 1000 Hz to beyond 16 kHz. The Bi-Radial compound flare configuration maintains precise control of the horn's wide 100° x 100° coverage angle. Since this angle is identical to the coverage angle of the low frequency driver at crossover, the transition from driver to driver appears seamless and the monitors present a fully coherent sound source.

And the Bi-Radial horn's performance advantages aren't limited to just beamwidth control. The horn's rapid flare rate, for instance, dramatically reduces second harmonic distortion and its shallow depth allows for optimal acoustic alignment of the drivers. This alignment lets the monitors fall well below the Blauert and Laws criteria for minimum audible time delay discrepancies.



Acoustic alignment of drivers (4430)

The practical benefits of the Bi-Radial horn design include flat frequency response and remarkably stable stereo imaging that remain valid over a wide range of listening positions. The design also allows considerable latitude in control room mounting. Finally, the flat on and off axis frequency response of the horn means that less high frequency equalization will be required to match typical house curves.

But while the Bi-Radial horn offers outstanding performance, it's only part of the new monitors' total package.

Extended Response in a Two-Way Design

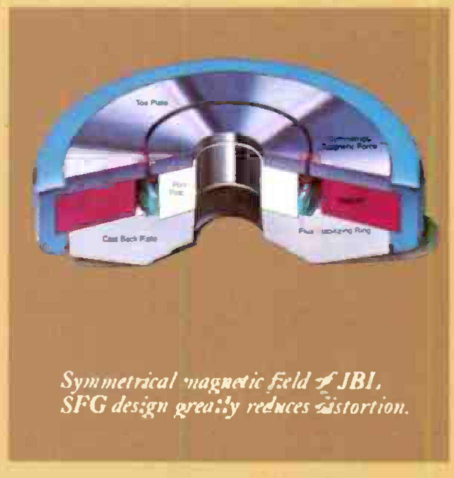
Coupled to the horn is a new compression driver that combines high reliability and power capacity with extended bandwidth and smooth, peak-free response. The driver features an aluminum diaphragm with a unique three-dimensional, diamond-pattern surround.[1] Both stronger and more flexible than conventional designs, this surround provides outstanding high frequency response, uniform diaphragm control, and maximum unit-to-unit performance consistency.

JBL's diamond suspension diaphragm combines performance with reliability.

To ensure smooth response to the lowest octaves, controlled midband sensitivity, extremely low distortion, and tight transient response, the Bi-Radial monitors also incorporate the latest in low frequency technology. The loudspeakers' magnetic structures feature JBL's unique Symmetrical Field Geometry (SFG) design to reduce second harmonic distortion to inconsequential levels. Additionally, the speakers utilize exceptionally long voice coils and carefully engineered suspension elements for maximum excursion linearity, and complete freedom from dynamic instabilities for tight, controlled transient response.

Blending the Elements— The Dividing Network Challenge

Tailored to the acoustical characteristics of the Bi-Radial monitors' high and low frequency drivers, the dividing network provides the smoothest possible response over the widest bandwidth while restricting any anomalies to an extremely narrow band. During the network's development, JBL engineers paid considerable attention to on-axis, off-axis, and total power response. As a result, the electrical characteristics of the network are optimized for flat response over the monitors' full coverage angle.



Symmetrical magnetic field of JBL SFG design greatly reduces distortion.

The network also provides equalization of the compression driver for flat power response output. This equalization is in two stages with separate adjustments for midrange and high frequencies.

Judge For Yourself

Of course, the only way to really judge a studio monitor is to listen for yourself. So before you invest in new monitors, ask your local JBL professional products dealer for a Bi-Radial monitor demonstration. And consider all the angles.

1. Patent applied for.

4430

4435

Specifications	4430	4435
Frequency response (± 3 dB)	35 - 16,000 Hz	30 - 16,000 Hz
Power Capacity (Continuous Program)	300 W	375 W
Sensitivity (1 W, 1 m)	93 dB	96 dB
Nominal Impedance	8 Ohms	8 Ohms
Dispersion Angle (− 6 dB)	100° x 100°	100° x 100°
Crossover Frequency	1 kHz	1 kHz
Network Controls	Mid Frequency Level High Frequency Level Switchable Bi-Amplification	

JBL Professional Products Division

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Figure 1. The Studer series 900 broadcast production console.

modules. A direct output is available from each input module, and sophisticated internal console switching for meters and monitors is provided.

These specifications evolved in the studio for maximum ease in the three major areas of multitrack music work: original tracking, sync overdubbing, and, of course, mixdown.

Because many users of today's multitrack studios entered the field after the introduction of this type of console, there sometimes is considerable confusion concerning the actual shape of a modern console. There is confusion concerning such things as the relationship between the number of mixing buses on a console and the number of outputs (direct) from that console, or between the number of outputs and the maximum number of tape monitor/meter channels available. Because it seems that the maximum numbers of these specifications are continually increasing (more buses, more channels of monitor for dual machine sessions, etc.) in the music recording studios, it is important to remember that other production users may not at this time require all of these maximum numbers.

Unfortunately it is a fact that console designs regardless of basic size have, within a series, tended to be either of one type above or the other, and as of now these two types (general purpose and multitrack) have diverged so far as to have left a considerable hole between them. Hence, for broadcast production use, many consoles are either too simple or too music/recording specialized. The broadcast production user must often either adapt an unsuited music recording console to the need or sacrifice features for certain practical considerations.

LOCATION OF CONTROLS

The typical multitrack console has evolved into a rather deep (front-to-back) size due to several factors, including more elaborate equalization facilities, the space needed for in-line monitor controls on the input modules, more bus assignment buttons, and the need to retain a low profile for good aural and visual "sight lines" in the control room. Controls are then placed at increasingly distant spots from the operator based on their frequency of use in *music recording session* practice. Indeed, some controls used only in broadcast may be completely absent. Unfortunately, the frequency of use may be considerably different in a production room where the use of the multitrack recorder is only one of the possible normal procedures.

IN-LINE MONITORING

The in-line monitor section mentioned above is a rather clever solution to a problem which arose early on in recording. As the number of input channels and tape channels began to increase greatly, consoles with separate multitrack monitor sections became increasingly long. This was more or less of a problem depending upon the division of responsibilities between engineer, producer, and assistant engineer, but sheer physical size was definitely becoming an operating problem, wheeled chairs notwithstanding! The addition of a monitor section physically folded back onto the input modules provided a greatly-reduced console length. However, since the introduction of the in-line monitor, other changes such as the use of the dual-channel module (two monitor channels in the width of one input channel) and the emerging demand for independent equalization on the monitor section have made the so-called side monitor again popular in many applications. In particular, the in-line monitor design has had a special impact on the broadcast production user, since the inclusion of this special section forces the production user into the particular range of a manufacturer's consoles which are designed primarily for music recording studios. Often, many of the desirable features the broadcast user may want and need are available from the maker on the "broadcast" models but not on the multitrack models from which he must select

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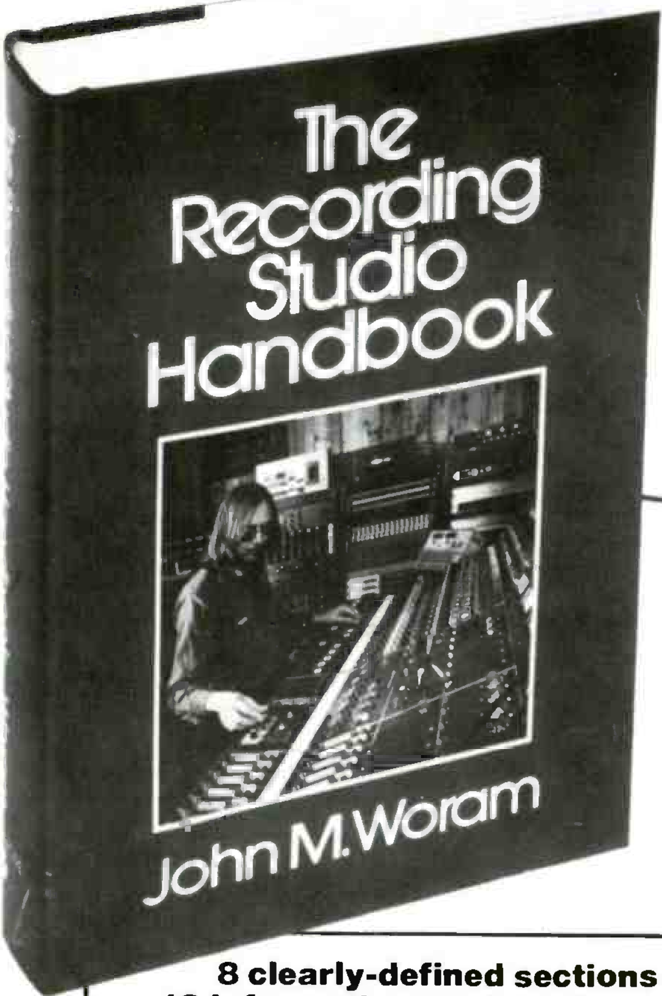

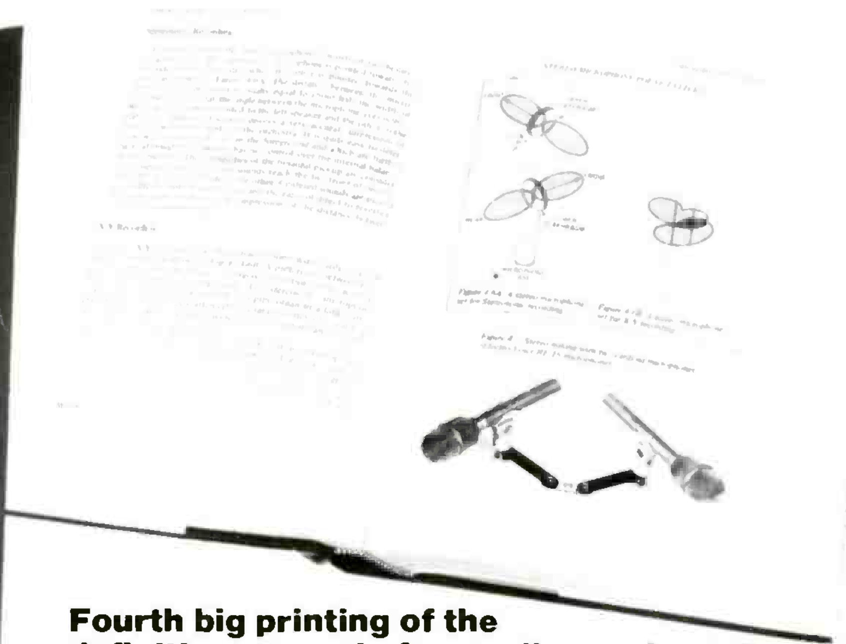

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for his multitrack production room. Or worse yet, the manufacturer may really only make multitrack studio consoles which are then changed in minor ways for broadcast production use.

INPUT MODULES

While the broadcast production user may require a variety of input modules, including stereo input modules, these are rarely offered on the typical multitrack music console.

The overall ergonometric design of the typical multitrack music console is for a very limited range of functions. The distance and relationships between operating controls, the relative positioning of metering and the overall physical areas of the controls are seldom optimized for broadcast production use. However, many other features, such as the basic low profile, the extensive equalization, metering and the auxiliary outputs are well suited to production.

It might be perfectly adequate to plan a given broadcast multitrack production room around an 8 bus, 24 track monitor console with direct outputs available. However, this type of console can be difficult to locate in the music recording console field these days except on a custom basis. Customers would be urged to "buy the full 24" when they might not need it. Similarly, it can be difficult to secure a top quality 8 bus, 8 track monitor console for production without ending up with a 16 or 24 track frame and a lot of blank panels.

It is thus proposed that although modern broadcast production studios may require 8, 16, or 24 tracks of recorded audio, the associated console designs are not necessarily best derived from recording studio consoles with the same number of tracks. The increasing sophistication of techniques points towards recording-type consoles, but with significant differences in the mechanics and audio architecture of the design.

One attempt to resolve this problem of making available features of both conventional and multitrack based consoles for broadcast production is to be found in the new Studer 900 console series described below.

The emphasis in the design of the consoles is on providing a mixing system that can be configured for both types of applications discussed above, utilizing standard modules based either on a conventional or multitrack monitor type format. An additional emphasis is placed on making a system that fulfills custom type requirements in recording *and* broadcast without requiring custom metalwork and frame design. Some basic considerations and specifications of the system are given below.

Mechanical Considerations—The basic low profile of the console allows for either multitrack or production control room use. The frame itself is modular, allowing for many variations in size within the same series. This means that designs from 12 inputs and 2 outputs to those with 50 inputs, 24 buses, and 24 track monitor can be achieved economically within the same system. In a multi-console facility, consoles of widely different applications would use the same basic spares. In addition, the middle configuration consoles so important in broadcast production (4-16 buses, with or without multitrack monitor) are an integral possibility within the system, not just "odd" sizes stuck between two distinct console lines.

For uses in both very large recording versions and remote vans, the compact basic module width of 40 mm retains manageable size. In addition, master output modules and monitor modules are dual density, allowing for very compact but still highly accessible units. Various combinations of metering are available including very compact VU or peak reading bargraph types also mounted in the dual-density arrangement. In multitrack versions, the track assignment buttons share the overbridge with vertical bargraph metering. Track metering is thus placed conveniently over the associated monitor/output sections and the console depth is reduced by the positioning of the assignment buttons.

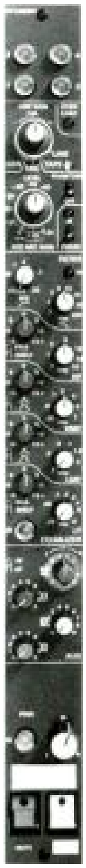


Figure 2. A mono mic/line input module.

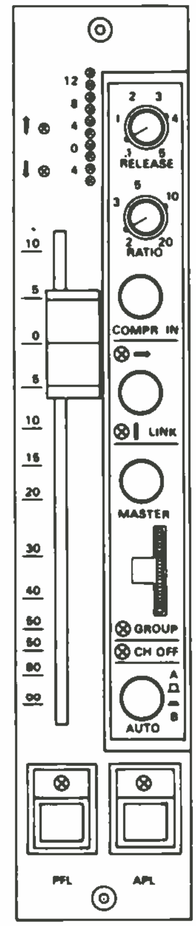


Figure 3. The VCA fader/limiter module.

Electronic Considerations—Any new generation of consoles must be aimed at performance compatibility with digital recorders and transmission systems. In addition to newly designed minimum path signal electronics and such options as transformerless microphone inputs, other aspects include transformerless tape line returns and a newly designed VCA system (described further below) for optimizing noise considerations.

Because of the wide variety of application of the series, the module selection is particularly diverse.

Input Modules—The basic Mono Input Unit has four inputs. These include MIC, which is actually a variable level input for microphone to line level signals from −70 to +20 dBu. The input is floating and balanced with switchable phantom powering and may optionally be transformerless. The next input is LINE, which is a transformer coupled line input with a variable trim. The third input is TAPE, which is the primary signal input for mixdown application and is transformerless and balanced. The final input is GEN, which is an auxiliary tone insert input to the individual channel.

The equalization available is quite comprehensive but compact due to the use of switch/pot combinations and push-pot combinations for many functions. The two sections include an HP/LP filter section with switched bypass for each filter, which has a 12 dB/octave slope and variable frequency. The second section is a four-band semi-parametric equalizer with shelving or peaking selection on the extreme frequency bands, and a choice of Q factor on the inner frequency bands. Of course, boost/cut amount and frequency are continously variable.

The module has the provision for stereo or quad panning and a direct feed switch. Auxiliary output feeds from this module include three mono auxiliary outputs, each switchable pre- or post-fader, and an additional stereo auxiliary output. On some console versions, an additional four auxiliary outputs will also be available on the upper switching modules. Other features include a silent mute and a user-assignable switch with LED indicator. An internal preamp overload indicator is also included. The solo buttons commonly found for channel solo are mounted external to this module.

There are several Stereo Input Modules available with features similar to the basic Mono unit. These include a Stereo Microphone Input unit, with equalization, a Stereo High Level Input unit and a Stereo High Level Input unit without equalization.

Switching Modules—On console versions with up to four buses, the input module contains all the necessary assignment switches. On larger versions, this switching is continued up vertically on switching modules in the overbridge section. Even in multitrack uses, this format helps to easily identify buses 1-4, which may be used in special mixdown options.

Fader Modules—Because these units match the various other modules for functions and because of the availability of the VCA system, there are a number of different fader modules, including those which contain master output line amplifiers, part of the very compact design of the console system. The basic single knob fader may be mono or stereo control. Below each fader are two pushbuttons to control the PFL/channel monitor solo and APL (after-pan listen)/channel positional solo functions. In addition, there is a separate monitor section solo function on multitrack monitor versions.

The dual fader unit corresponds to the stereo fader, but has a separate fader control for each channel. Another version is the master fader which contains faders, line amplifiers and an optional VCA-based limiter.

VCA System—A newly-designed VCA system is available in the console series. This new VCA is designed and built by Studer and has at least two interesting aspects. 1) Excellent long-term temperature stability makes periodic adjustment unnecessary and accordingly, there is no trim pot for this re-alignment; 2) The VCA operates in either class A or

class AB mode, depending upon an applied control voltage. In the design of the VCA fader modules, these possibilities have been utilized to provide the following action: depending upon the applied input level, the optimum operating point between class A and AB is selected. The mutual dependence of noise performance-versus-distortion is shifted in the direction of large dynamic range if a very low level signal is applied, and in the direction of minimum large signal distortion in the case of a high level signal.

The VCA fader system is available for external control by any DC control automation system, and additionally provides the following features: Up to ten DC control groups can be created. If no group is selected, but both the MASTER and LINK pushbuttons are selected, there will also be created a separate Stereo Group independent of the other groups, with the channel to the right. Muting control of channels is available.

The built-in limiter/compressor also utilizes the VCA as the control element. The ratio can be adjusted from 2:1 to 20:1. It increases continuously from 1:1 to the selected value in a soft transfer zone. The release control allows adjustment of the recovery time which is also program controlled depending upon the initial level.

Monitor Modules—Tape Monitor modules, when specified in a system, are dual density (two channels/40 mm) units with normal monitor functions plus three-band equalization, four auxiliary feeds, and provision for reversing the functions between the rotary monitor level control and the linear fader of the master output. A monitor solo function is also included. The monitor section can be group source switched between console output or tape output.

Remix Mode—A console master Remix mode is provided on multitrack versions which will switch channel inputs to tape and switch mixdown and multitrack metering. In addition, mixdown can be achieved either through the channel section or through the monitor section. On non-multitrack monitor versions, the Remix function still exists to switch channel inputs to tape.

Studio Monitor & Talkback Unit—This module contains switching and talkback control for studio monitoring from many sources, as well as bus amplifiers for solo functions within the console. Extended talkback facilities can be supplied for remote van applications or others where needed.

Control Room Monitor Unit—Up to 15 monitor sources may be selected. Various housekeeping functions, such as monitor level, phase reverse, dim, headphone jacks, etc. are included in this module.

Auxiliary Master Unit—Each group of four auxiliary outputs has an output master unit including high-pass filter equalization and mute and solo functions.

Other Modules—As part of the system for dual use, other modules include Direct Input to Auxiliary and Direct Input to Monitor modules, special Switching Modules, including master clear feed versions, Input Selector Modules for multiple source selection beyond that already available, and a set of Remote Control & Timer Modules.

SUMMARY

Even though current multitrack console design practice has resulted in an audio and physical console architecture that is overly specific to a single use, it is possible to integrate the various requirements of multitrack music recording and advanced broadcast production in one console system which can then fulfill requirements in both of these overlapping areas. ■

NIGEL BRANWELL

Dynamic Processing—Part II

Author Branwell continues his discussion on dynamic processing, focusing in on the concept of multiband processing.

IN PART I of Dynamic Processing, multi-band processing was touched on briefly as a method of achieving greater gain reduction than broadband processing will allow due to modulation effects.

The concept of multi-band processing is not new; as far back as the early seventies, Audio and Design introduced its first band-splitting systems and other systems may, indeed, have preceded these.

MULTI-BAND IN PRODUCTION

Conventional compressors and limiters attenuate equally throughout the audio frequency range as the gain reduction is triggered by a signal at *any* frequency exceeding the pre-established threshold. Hence, high level, low frequency signals (e.g., organ, drums, timpani, or bass) are frequently impossible to compress or limit without also modulating high frequency program material or ambience. This can be noticeably degrading.

The ear detects gain change by its modulation of other signal content, particularly ambience. Normally, using close-microphone techniques and compressing individual instruments (or groups of instruments) robs the ear of an external reference so that considerable compression can be achieved without being apparent. However, there are many times when such a technique is not possible: for example, problems arise when transferring the finished recording from one medium to another (e.g., tape-to-disc, cassette, optical film track, video tape, and broadcast transmission).

In the case of disc, difficulties are often found in the low-frequency region, giving rise to tracking problems. For high speed duplication of cassettes, it is usually the sibilant area that will spoil an otherwise good copy. In optical track, video tape and broadcast transmission, it can be a combination of the above plus the restrictions imposed by the limited dynamic range of the new medium compared to the wider dynamic range of the original. In the case of broadcasting at least, there is often the requirement to maximize apparent loudness and impact. Bear in mind that at this stage, a final balanced program is being processed—usually the result of many hours of hard work by producer, engineer, and artists resulting in what they ideally hope the listener will want to hear. It is the production engineer's job to see that it gets over to the final medium with the minimum change or losses.

Very often, in attempting the transfer, the choice is to either lower the modulation level on the new medium or insert fixed

Nigel Branwell is vice president of Audio + Design Recording Inc.

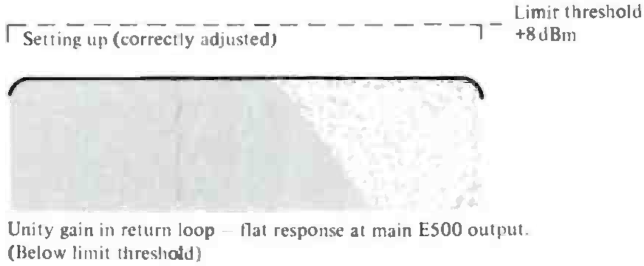


Figure 1. Here, the high-pass filter has been used to select the low-frequency content (shaded) which is routed to the limiter. After limiting, it is re-combined to provide a flat response at the main output.

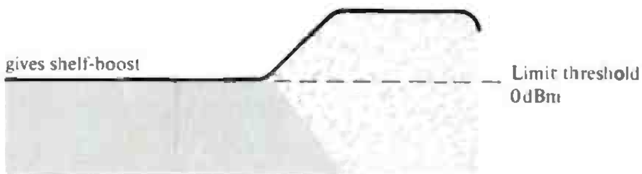


Figure 2. The output response, with the limit threshold set at 0 dBm.

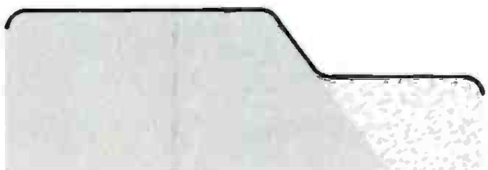


Figure 3. The effect of gain in the send/return loop.

attenuation (equalization) in the troublesome areas—which permanently degrades the signal. The ideal answer when minimal modification of the original is desired is *Dynamic Equalization:* this is the selective and momentary attenuation of frequencies determined by program level at *those* frequencies only. Note that this is *not* the same as having a frequency-selective "side-chain" (or control voltage) in a limiter, where the whole program is still modulated and is thus only really useful on individual tracks.

The problem portion of the signal must be split off from the rest of the program and then added back at a later stage after processing. Equalization is then present only *above* certain levels (using a compressor or limiter) or *below*, using an expander (or gate). At all "normal" levels the system response is quite flat; the program is therefore *momentarily* modified to suit the new medium, as necessary.

While the advantages of dynamic equalization have been appreciated for some time, the circuitry required to split the bandwidth in such a way that it can be accurately reassembled is complex—especially when coupled with the need for versatile selection facilities. Taking Audio & Design's Transdynamic Tri-band processor as an example, the unit first splits the incoming signal into its hi pass band pass/lo filter combination (6 dB or 12 dB per octave) as defined by the turnover points selected in the crossover and provides a line level send to, and return from, any external audio processor for each of the three frequency bands. If the send is directly linked to the return (unity gain), the signal appearing at the main output will be identical to that at the input and any action taken in the filter sections has no effect. Should the loop be broken, the unit then acts as a static equalizer, providing high- and low-pass and band-pass filters. Providing some fixed amount of

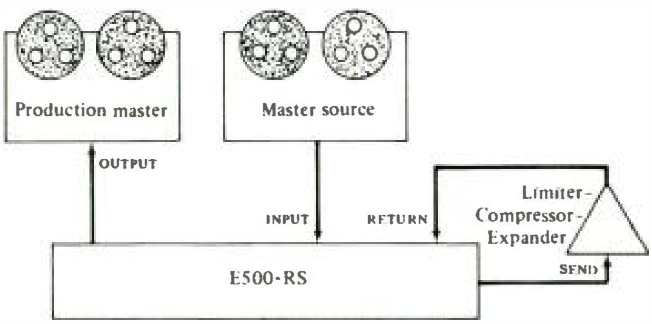
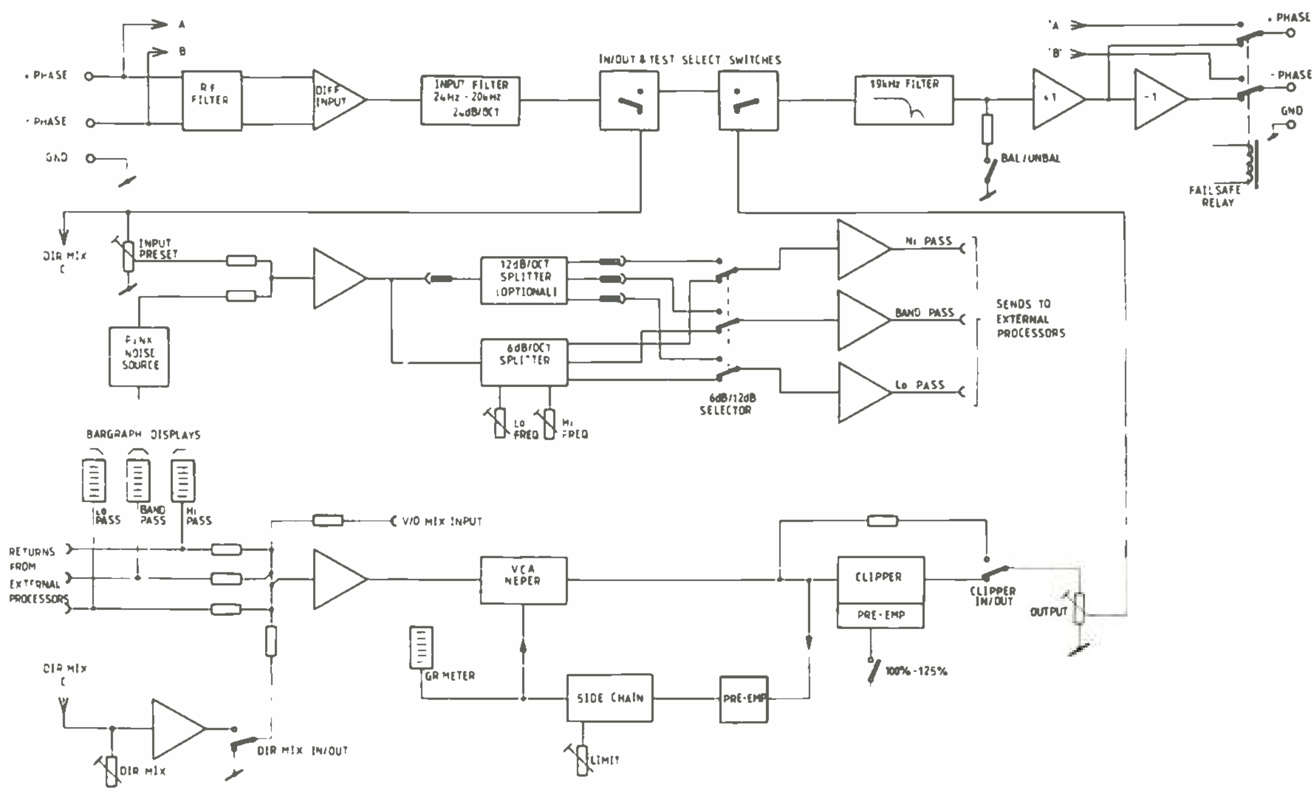


Figure 4. The E500 Band Processor integrated into an operational system. The output from the master tape is fed into the main input, and the main output is connected to the copy tape machine. The external send/return connections are routed to an external compressor/ limiter, etc.

Figure 5. The Transdynamic Tri-band Processing System.



attenuation in the loop (e.g., 10 dB) will result in a shelf-type equalization (with 10 dB shelf). These are passive uses of the device. Its real benefits can be realized by inserting a level control device in the external loop (limiter, compressor, expander) of any or all three bands. Equalization will now be dependent on program level and the amount of attenuation *in the selected region only* will be determined by the program level relative to the threshold setting of the external processor. In the case of the Transdynamic, the crossover filters are variable between 75 Hz and 1 kHz on the Low-Pass-to-Mid section, and between 1.1 kHz and 15 kHz on the Mid-to-High-Pass section. With a compressor-limiter-expander on all three bands, significantly higher level can be achieved while retaining a flat dynamic response or, if desired, introducing dynamic equalization to suit the following medium (more on this when discussing broadcast applications).

Note that the above applications requiring the compressor-limiter-expander as an external processor are, of course, solutions to *level* control problems of one type or another. It is equally valid to use a multi-band system to define a group of instruments or voices for external processing via time delay (phase, flange, ADT etc.) or other effects equipment. It is even possible to add echo equalization and double tracking *post* production and thus add *selectively* to a master tape.

MULTI-BAND IN BROADCASTING

Here, the same principles of multi-band processing as discussed above are used. The goals however may be quite different. Broadcast stations, particularly radio in the United States (and perhaps soon in Britain and Europe), tend to concentrate on particular types of program material (format) and seek to engineer a station "sound" that will help put them ahead in terms of the competition (i.e., audience rating). Often this has turned into a question of who can be the loudest station in the coverage area for a given format which has in turn led to "loudness wars" between competing stations; fortunately, increasing evidence supporting the theory that loudness per se does *not* lead to increased listeners is mounting.

Broadcasting is about communication (as is the entire audio industry); to communicate, it is necessary to remain in touch with the listener. Audio processing is all about communication; it seeks to sensure that the listener is A), attracted to the station as he flips across the dial, and B), that the signal is maintained at the highest quality while remaining at all times intelligible to the listener. This means having a sufficiently high average level in order to stay above the ambient noise in the listener's environment with low level signal still audible at typical listening levels. It could be that the same program is transmitted on both AM and FM, being processed differently to meet the requirements of the two transmission mediums (given the current poor quality of AM receivers), and the restricted listening dynamic range available while travelling and the wider dynamic range in the listener's home.

Multi-band processing, in this case with processors operating on each of two or more frequency bands, will increase energy levels without the modulation effects of a broadband processor, distributing the frequency spectrum dynamically according to the format, the transmission medium, and probably the program director too! As before, the process of band-splitting, the quality and flexibility of the band compressors, and that of the final peak limiter will determine the overall sound quality that can be achieved.

A three-band system is more than adequate for purposes of obviating modulation effects; high compression can be achieved for AM (more so if the external processors have auto release networks for AGC action) and overall gain in excess of 16 dB can be successfully processed on 6 dB/octave crossovers without modulation problems. The use of 12 dB/octave

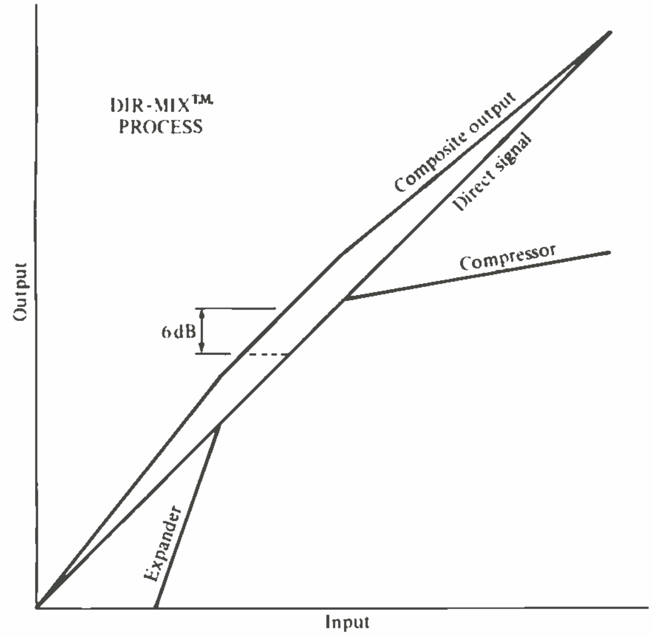


Figure 6. The DIR-MIX™ process for wide range dynamic processing using a Transdynamic Processor and three F760X Compex-Limiters.

filters will allow even greater density but are of more use where overall bandwidth is severely restricted (e.g., 5 kHz in the UK on the AM band). Other than modulation effects, the main problem with any form of compression is that source noise is increased by the gain set under quiescent conditions in the compressors; in any well-designed compressor, source noise will always exceed system noise (even from digital tapes) assuming 10 dB overall increase in average level. Expanders can help here by operating below wanted signal to attenuate each band progressively as level falls. Given sensible cross-over points in the filters, expanders will work well on orchestral music and are ideal for use in AM applications where a substantial amount of compression is often used. This is a form of dynamic equalization—as the high frequency content falls, or is absent, the effect will be that of a 6 dB/octave (or 12 dB/octave) shelving filter acting on low amplitudes only.

The gain change in multi-band processing for broadcast use should be identical throughout the input range in each band to retain a flat response *dynamically*. In practice, the gain change or compression will vary from moment-to-moment in each band dependent on the program energy levels at the time. If a sensibly flat response is required, ratio, threshold, release, or crossover point of the filters can be adjusted to spread the compression over comparable ranges in each band section. Gain change in each band is then closely matched to retain a flat dynamic response. Note too that *fixed* parameter multi-band systems will, more often than not, act as dynamic equalizers to the degree that gain change varies between bands (the more bands, the worse the problem). This can lead to a total corruption of internal balance, a flattening of the sound and worst of all, listener fatigue. Some systems also resort to band coupling to reduce distortions of dynamic response. Here, the result will be increased susceptibility to modulation problems, thus defeating the original purpose of band-splitting.

While subtle variations in dynamic response can lead to an enhancement effect on the signal (apart from any increase in loudness), it can be easily overdone; a degree of increase in low level high frequency content relative to high-level mid/low-frequency content can be subjectively attractive. Many kinds of dynamic response can be shaped; by increasing gain on high- and low-frequency sections and applying greater compression so that the high level output remains flat, low level signal will increasingly be lifted as signal level falls, giving a dynamic Fletcher-Munson loudness response.

By selecting a limit or tight compression slope for the mid-frequency section and a soft ratio on low and high frequencies, presence signals can be maintained at high level until full recovery, while LF and HF is reducing. This gives a dynamic presence peak at mid amplitudes that progressively flattens at high and low levels. When applied to the 3 kHz frequencies, this can be useful where bandwidth is restricted to 5 kHz or 11 kHz (AM). For subtle operation on wide dynamic program, soft ratios (1.5:1 or 2:1) will be found the best; using tighter slopes (5:1 or 10:1) will maximize loudness.

Using one external unit to provide selective HF or LF limiting; or two for selective HF and LF control prior to master limiting, excessive HF or LF (or both) signals that would normally trigger limiting are momentarily attenuated while the master limiter, if operating at all, responds to the mid-band signal.

With this approach, the signal retains its full dynamics with only occasional and momentary modification, which is totally inaudible except on A/B comparison under hi-fi monitoring conditions; yet the transmitter output can effectively be increased by as much as 6 dB.

The selection of de-emphasis in the master limiter will act as a peak indicator of excessive HF content which can be dynamically controlled in the external compressor. Thus, selective HF compression can be adjusted to give dynamic de-emphasis given the selection of a comparable turnover frequency to that of the transmitter pre-emphasis, on the MF crossover.

For wide dynamic formats, discussed in Part 1, the direct signal can be paralleled with the multiband processed signal

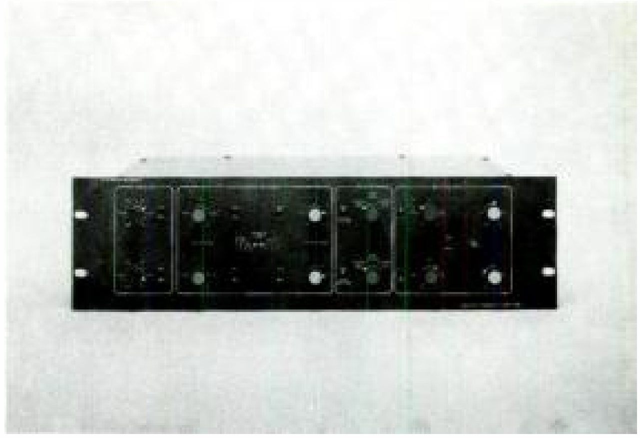


Figure 7. The Audio & Design Transdynamic Processor and the model E500 Band Processor.

producing the effects of voltage addition. Mixing the compressors in at low level (some 25 dB down with reference to peak signals—which would mean 25 dB-plus compression at high level), the compressors predominate at low level, adding gain, while as level increases, the direct signal increasingly takes over and predominates at peak levels. Added equally at low level, the gain will increase 6 dB returning to unity at peak levels. FIGURE 5 shows the effect of adding a compressor/expander combination slope to the direct signal level. The advantages of this system outweigh any disadvantages substantially—with the compressors added 25 dB down on the main signal, any noise or distortion is a further 25 dB down on normal. Any ratio selected is further softened (2:1 becomes 1.3:1 approx.) and compression is spread quite smoothly throughout the input range so that internal dynamic relationships are more accurately retained. This type of system works very well for wide dynamic program retaining the crystal clarity of the upper dynamics without the lower level compression being noticeable.

As an experiment, this system was tried on a sizeable orchestra. Having been recorded flat, the tapes were then replayed using the system described above. Not one musician realized that his/her instrument had been modified in any way. It works and is a system that should be of interest to classical and jazz stations as well as an operational balance aid when recording a difficult wide dynamic program.

The advent of digital mastering is an interesting prospect for use of a quality multi-band processing system. The major problem, other than modulation effects, is the way source noise is increased with compression. With a digital recording capable of accommodating the widest input dynamics, the ambience of the studio will always mask any electronic noise. Subsequent processing through a multi-band system of suitable quality will only increase ambience rather than electronic noise.

Audio & Design produce a number of systems employing multi-band techniques. The latest is the Transdynamic tri-band system, some features of which have already been described. Other features include a high-quality wide-band master limiter and optional clipper following recombining of the three frequency bands, selectable 0, 25, 50 or 75μs pre-emphasis in the control circuit rather than the audio path, adjustable assymetry for AM applications, optional bandwidth contouring filters at the main inputs and outputs. LED bargraph metering indicates gain reduction in the master limiter, clipping action (if switched in) and return levels from the external processors. A pulsed pink noise generator facilitates initial calibration of the system.

The E500 Band Processor's most pertinent features are a Selective Band Processing section on each of the two channels comprising two fourth order filters, one High Pass and one Low Pass—both continuously variable over a frequency range of 100 Hz to 10 kHz. Each filter can be cancelled to allow operation individually or together when a mid-band frequency component is selected for processing. ■

VLADIMIR NIKANOROV

A Computerized Console For Radio Broadcasting

Automatrix is a new broadcasting console which uses modern technology to produce professional-quality audio, combining the features of manual and automated operation.

CONVENTIONAL "AIR" CONSOLES for radio broadcasting are a product of long established broadcast studio procedures. Generally there are two types of studio operations in which a typical broadcast air console plays the main role. One is a manual operation, another is an automated operation.

In the manual operation, the console is basically a single-output summing amplifier. Remote start-and-stop controls are often a part of the console, as seen in FIGURE 1. Turntables, open reel playback machines, cartridge playback machines, and microphones feed the individual inputs of the board. The console operation opens and closes input faders and activates the sources by the remote control system on the board.

In a typical automated operation, an air console is basically used as a microphone mixer to control levels and as an audio monitor. The console provides live audio source to the automation system, which in turn performs as a summing amplifier and as a programmable switcher, alternating the audio sources in the desired sequence (FIGURE 2).

Vladimir Nikanorov is technical director of Bonneville Broadcast Consultants, Tenafly, NJ.

Both systems have their advantages and shortcomings. The manual operation requires constant and intense attention. Routinely, records are spun, tape decks are fired up, and announcements are made, usually by the same person who is also expected to pay attention to levels on VU meters and to read the modulation monitors. The quality of the show is greatly dependent on this operator, whose skill and individuality really makes this type of operation an art. Some advantages of the manual operation are flexibility in selecting audio sources, spontaneity, and sometimes the involvement of the announcer's personality.

However, when the operation is partially automated, the intensity of the manual real-time show can be diluted over an extended period of time. This may remove the "art" and individual aspect (along with a great number of errors) from the operation.

The fully-automated broadcast operation usually uses recorded announcements, and no provision is made for handling live audio or for playing gramaphone records. Typically, the tape reproducers become part of the automation system, which consists of huge 19-in. racks full of equipment. Open reel and cartridge players are hardwired to the system switcher whose function is to connect one source or another to a program amplifier. The output of the amplifier feeds the air board. The switcher normally is controlled by a microprocessor, which in turn is fed with computer data.

Selecting an audio source becomes a slow and premeditated process which usually requires a separate operator and, in many cases, makes the manual operation appear more desirable

because of its flexibility.

Despite the fact that automated broadcasting relaxes the on-air studio operation, it is potentially hazardous because of the system's structural design. For example, if a switcher or program amplifier fails, then the station is off-the-air regardless of the state of the audio sources.

THE AUTOMATRIX AIR CONSOLE

A desire to achieve the flexibility and cost-effectiveness of manual operation, together with the "error free" advantages of automated operation was the design philosophy behind the Automatrix computer-assisted air console.

The heart of the console is the Audio Main Frame Unit (AMU), which—as in a regular air console—is nothing but a complex summing amplifier with a single audio program output. The audio output is fed to outside circuits: radio broadcast processing equipment, exciter and transmitter. Typically, 16 stereo sources can be connected to 16 individual inputs of the AMU. The number can be extended up to 32 sources.

The Machine Control Unit (MCU) provides automatic start-stop functions for the audio sources. The Board Module Group (BMG) contains monitors, VU and PPM meters, and a manual remote control switchboard for the 16 sources.

So far, the block diagram is quite similar to a typical manually-operated air console. However, similarities of the console structure with conventional designs start and end with the fact that it is made to deal with audio sources. Unlike most of the air consoles, it functions automatically, taking commands from the System Control Unit (SCU), the next block on the block diagram.

The SCU feeds 8-bit data to the D/A converters on each individual audio input card. The D/As supply variable DC voltage to the VCAs (Voltage Controlled Amplifiers) on the audio input cards, which in turn change the audio levels. The levels can be brought to a predetermined value or completely turned off.

The next and the last block is the Programming Computer Group (PCG). The PCG computer controls the SCU functions over a desired period of time.

AUDIO MAIN FRAME UNIT (AMU)

The AMU contains 16 or 32 stereo cards. A block diagram of a typical audio input card is on FIGURE 3.

The input is active balanced and is based on an instrumentation amplifier feeding two identical VCAs: one for program and one for cue outputs. Audio level information for the VCAs is carried by an 8-bit data bus from the SCU to the D/A converters. The D/As are calibrated with a reference generator which makes possible precise level adjustments. The levels can be pre-programmed or changed by the BMG during the course of the program if necessary.

Outputs of the input cards are fed into the line amplifier which is terminated by a summing program amplifier on the program output card.

Start-stop of audio sources is done automatically in different ways, depending on operational requirements. An end-of-event circuit (EOE) senses the 25 Hz tone routinely used to stop-start open reel players or a 150 Hz tone for cart players. The circuit can be tuned either way.

Detection of the tone will cause the comparator to send a 1-bit digital signal to an addressable buffer circuit and on to the SCU via the data bus. Then, depending on what it was told to do previously by the computer, the SCU will react and activate the next desirable event.

Another way of controlling events is based on the absence of audio. In addition to feeding the audio program to the external broadcast chain, the left and right summing amplifier outputs are summed and rectified. The resultant DC is converted into a digital word which is constantly read and compared to a reference standard. If a no-signal condition is detected, the SCU will activate a new event. An optional alarm will indicate silence.

SYSTEM CONTROL UNIT (SCU)

A block diagram of the SCU is seen in FIGURE 4. The SCU is the intelligence (limited however) inside the console. It contains 4K EPROM (Erasable Programmable Read Only Memory), 1K RAM (Random Access Memory), the microprocessor, control, address and data buffer drivers and standard RS232 communication interfaces which are linked to the main Progamming Computer Group (PCG) and to the console's Board Module Group (BMG).

The heart of the SCU is the Microprocessor Unit (MPU). It manages the following operations:

1. Operations within the AMU, for controlling the VCA setting.
2. Operations within the MCU: machine starts and stops, level sensing and silence sensing.
3. Runs program which scans EOE (End-of-Event) buffers.
4. Reads A/D converters, starts A/D silence-sensing converters and checks status of RS232 interface registers.

Depending on the system operation software, which operates the EPROM (Erasable Programmable Read Only Memory), the MPU starts and stops sources selectively, sets up the levels, makes sure that audio of a source is actually at the system output and keeps in RAM (Random Access Memory) what should be done for the next events.

THE BOARD MODULE GROUP

A block diagram of the BMG is shown in FIGURE 5. This section of the system is designed specifically to provide the extra flexibility required for automated broadcast operations. The BMG consists of the following modules: a microprocessor-based console control module which can act independently or in conjunction with the SCU through the RS232 interface and a special functions module. This module and the MPU of the group act in the same manner as the SCU when it senses the levels and sets the sources on and off. The special functions module can be used for live cross fades or predetermined fades of any duration, which makes it possible for live cross fades or for example to play gramaphone records either in the traditional manual fashion or in automated mode.

One of the most important functions of the console control module is to convert the automated operation into manual. This is done by a single manual/automated push button. At the moment of the conversion, 16 pairs of push buttons on the front panel become start/stop remote controls for the audio sources. The Board Module Group also contains microphone modules, VU and PPM meters, earphone feeds, manual cue amplifier, network access module and telephone (patch) module.

THE PROGRAMMING COMPUTER GROUP (PCG)

A computer in the system is used as an extended intelligence linked to the MPU of the SCU. The computer, which is interfaced with the MPU by standard communication interface RS232, constantly updates the system and typically has a capacity of remembering about 10,000 events.

Standard operational software programs are written for the Apple II-Plus computer and is used with pre-edited events sequencing and record keeping. Alternate or extended computer systems can also be used, in which case the software and the interface should be modified.

BROADCAST PROGRAMMING

Conventional broadcast programs are based on an audio source schedule. This will include level, fade and cross-fade information, cues for live announcements, and assists for semi-live functions such as gramaphone record playing.

The programming is compiled by the main computer keyboard and CRT, located on the right side of the console. The on-air announcer has instant access to the keyboard which is in fact a part of the air console. This eliminates the additional operator normally required in conventionally-structured automated operations.

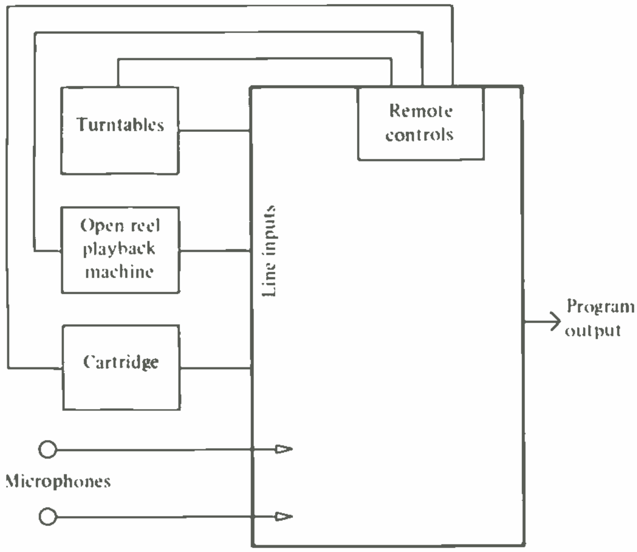


Figure 1. Block diagram of a typical manual operation.

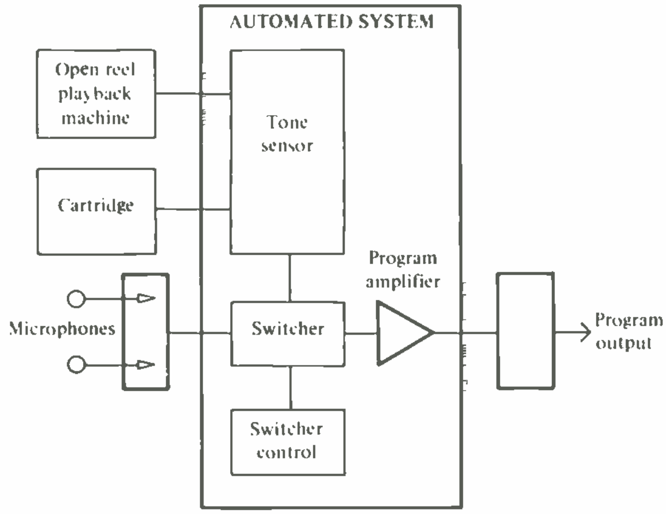
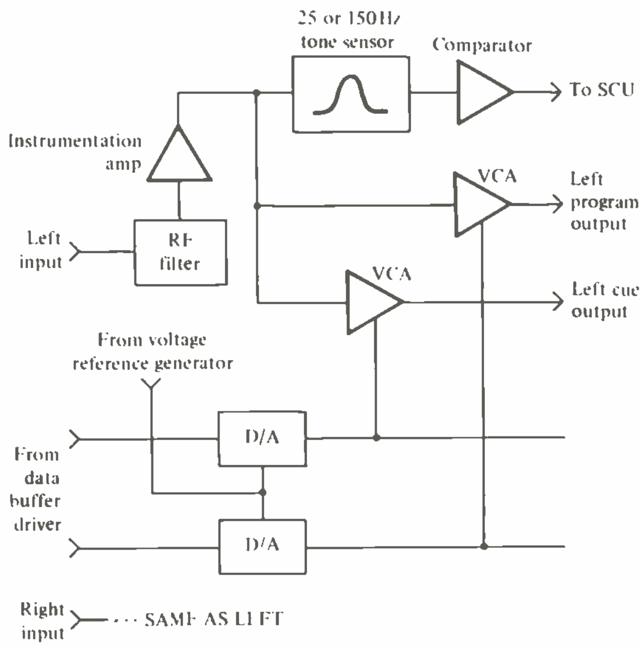


Figure 2. Block diagram of a typical automated operation.

Figure 3. The Audio Mainframe Unit (AMU).



The programming can be done either in English language (i.e., BASIC) or in code. The program, which controls the system in real time, is stored in the floppy disc memory and is transferred in increments to the console's MPU, which in turn will transmit the events instructions to the SCU drivers.

While transferring, the CRT can be used for a number of different purposes. For example, it can provide indications for up to 12 events in advance with a current event countdown or it can display an analog clock with local time (the analog clock is a standard feature of the Apple II-plus computer).

The disc's memory is also used to keep records of real time events which are then transferred to hard copy. The printouts are used for the FCC logs, billing, etc.

PHYSICAL DESIGN, QUALITY OF AUDIO PERFORMANCE OPERATION

Neither AMX console resembles a recording desk or a conventional broadcast console. They look more like space age office desks with lots of controls on the left, a computer keyboard and CRT on the right and a meter panel at the rear center. The desk provides a working space for the announcer/operator who, by the nature of the procedure, has to deal with a lot of paper work.

As mentioned before, the console is a modular structure. The modular philosophy was carried all the way through.

Each group of modules constitute a bigger module, practically an independent box. This allows the separation of audio and digital circuits, makes possible relatively easy expansion, simplifies service and is cost effective.

The Automatrix console is marketed by Aphex Systems, Ltd. The basic price of the model AMX 16 is about $15,000, computer excluded.

Great attention is given to the station's environmental circumstances. For example, both "high" and "low" ends of the balanced audio circuits pass through RF filters before entering or leaving the Audio Main Frame Unit (AMU) box.

The AMU is contained in one or two (depending on the model) standard 19-in. chassis which are mounted on the lower back of the deck.

The input and output connections are on a barrier strip terminal (about the only thing which is done conventionally in the AMX console).

The inputs and outputs are active balanced, bridging; however, external input-output transformer barriers can be furnished if desired. Input-output clipping occurs at +27 dBV, as referenced to 0.775 VRMS. For a typical broadcast input level of +8 dBm, the head room of the board is +19 dBm.

Signal-to-noise ratio is –80 dB, referenced to the nominal signal level of +8 dBm. THD at the nominal level is 0.01 percent. Frequency response is flat within 0.1 dB from 30 Hz to 30 kHz.

A number of the boxes containing primarily digital control circuits are mounted in a separate 19-in. rack. Among them are the System Control Unit (SCU), the Machine Control Unit (MCU) and the power supply. Actually, the Digital-Analog

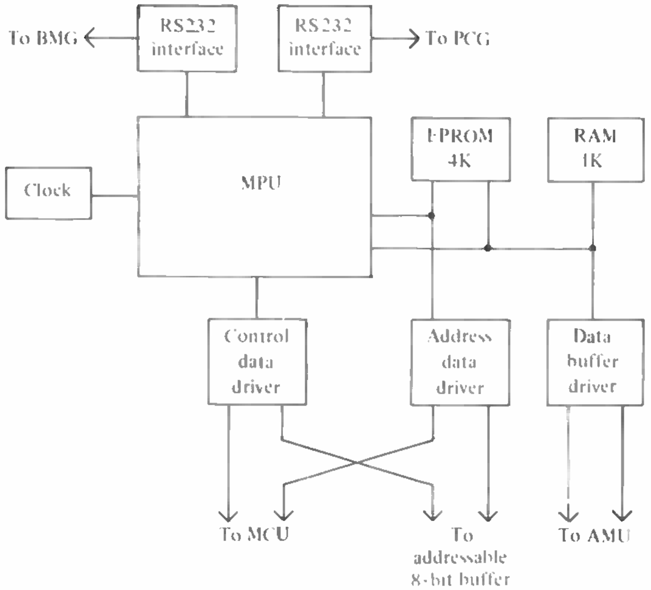
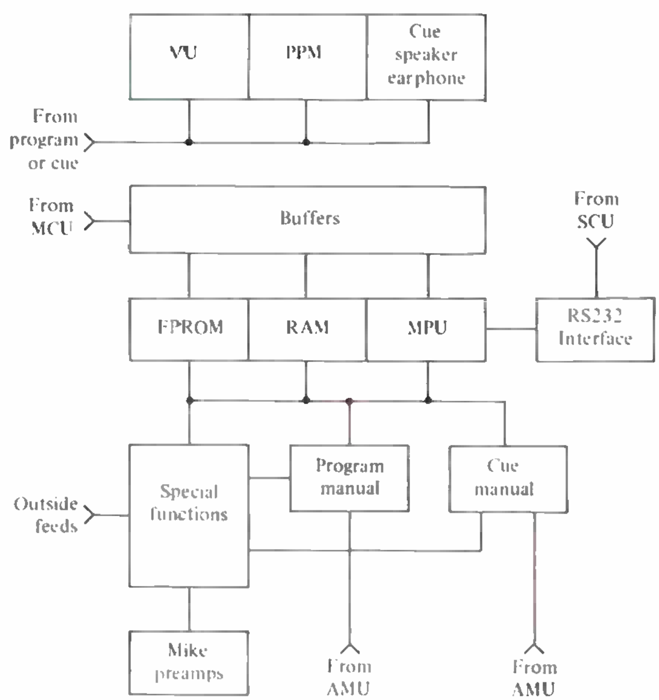


Figure 4. The System Control Unit (SCU).

Figure 5. The Board Module Group (BMG).



Module (DAM) and the Board Module Group (BMG) are the only modules that are plugged into the console mother board.

CONSOLE OPERATOR

When an announcer/operator sits at the console, he or she is confronted with just one push button. This button switches between manual and automated modes.

One of the drawbacks of regular automation systems, and therefore of broadcast formats which are run automated, is the difficulty of inserting anything within the preprogrammed flow.

This is especially true in the case of instant live announcements. Short of bypassing the system all together, the announcements must be programmed ahead of time—which of course takes away the spontaneity and value of timeliness. The AMX completely overcomes this type of drawback. When a need arises to go live or insert a new piece of programming into a regular format, the announcer simply pushes the manual/automated button.

The current event is halted by its dedicated remote stop button and the new program is started. The microprocessor of the Board Module Group remembers where the main program was stopped. It is activated by the manual automated push button and then switches all the source function controls to manual mode. However, the end-of-events circuits (EOE) remain activated. A switchboard of 16 or 32 pairs of LED-illuminated push buttons becomes the manual remote control board for the audio sources. If a source is activated, then the LED on its dedicated start button is illuminated. Previously-programmed operation will resume (if not re-programmed) right from the place where it was left off at the return of the button to the automated position.

The simplicity and quick switchover from automated to manual and back in combination, with the ability to set different levels with automatic fade and cross fade, allows the operator to do neat things with insertions and spots. For example, the console can be used for precise real time editing and processing of commercials on the spot.

Actually, the potentials of the console may suggest an entirely new style of radio broadcast production routine. A provision is made in the design for the production to be done while on the air. Cue circuits which duplicate the program chain can be used for production, while the main chain runs the regular material. Spots are known for bad recording qualities, and especially for discrepancy in levels. In the new style of production, the commercial can be auditioned, and the levels logged and changed when the spot is programmed at the computer.

The possibility of real-time preprogrammed edits or live changes of levels and instantaneous change from and to automated mode also opens new possibilities for formats based on playing gramaphone records

Normally the DJ is involved in many mechanical operations which consume his or her time "behind the scenes" and often gets in the way of the quality of the announcements. Automatrix can take over most of these time-consuming operations.

It will take care of starts and stops of the turntables and tape machines, give a cue and open the announcer mike, make the right choice at the right time (about the next source) and will allow the announcer to prerecord and properly insert commercials and portions of program which are difficult to do live. The disc jockey will be left with only the function of announcing, cueing the records and entering instructions in the computer.

CONCLUSION

A computer assisted console with excellent audio quality is not something unknown. The recording industry has used automated consoles for multitrack mixdowns for years.

In the radio broadcasting industry, many automation systems are nothing else more than glorified switching mechanisms, which quite often do not satisfy the main purpose of broadcasting, to produce high quality professional audio. ■

MELVIN C. SPRINKLE

An Artificial Microphone

With the use of an artificial microphone, obtaining Proof of Performance measurements can be as easy as one, two, three.

IN THE RULES & REGULATIONS governing standard broadcast stations, including those using the AM as well as the FM system of modulation, Uncle Charlie requires that every station make a Proof of Performance measurement. This consists of a number of tests made on the station's audio system and the transmitter.

Although the vast majority of stations originate their entertainment material from either disc or tape recordings, the Proof of Performance regimen requires that the station make an amplitude-frequency response using a microphone channel as an input.

This requirement presents some problems for those who lack proper equipment, but is a breeze for those with the right apparatus.

In order to make the measurement, a sine wave signal is fed into a microphone input—that is by far the most common method. The input signal is measured if the station has a sensitive AC voltmeter.

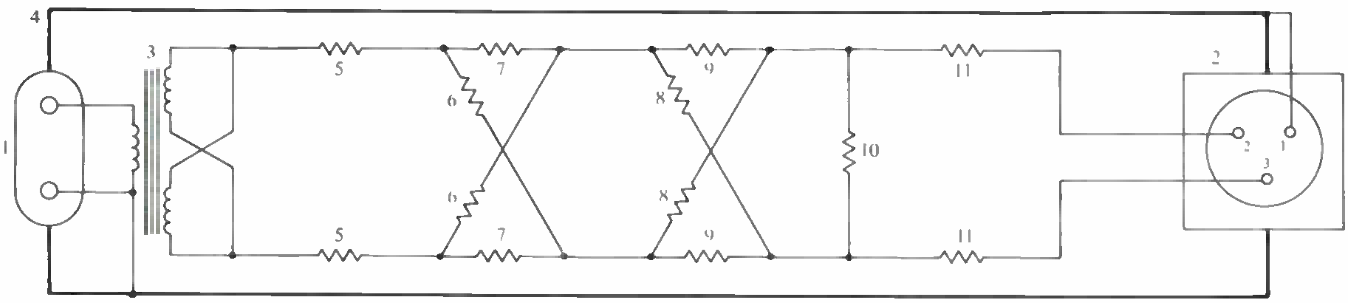
Unfortunately, however, this method is fraught with pitfalls and one is more than likely to wind up with the wrong answer for amplitude-frequency response.

There are several problems associated with this approach: (1) all microphone preliminary amplifiers have a *balanced* input and (2) a very minute signal is required to prevent overload and distortion in this first amplifier: a typical value is 1 millivolt. (3) The proper source impedance is not presented to the input stage.

Mel Sprinkle is a principal in Sprinkle Associates, Audio Acoustical Engineers

The vast majority of sine wave signal sources available on the market are eminently *unsuited* for use in a Proof-of-Performance measurement on a microphone input. In the first place, most of them are *unbalanced* and the direct connection of an unbalanced piece of equipment to a balanced circuit without an intervening transformer is a "no-no" in good engineering practice. It might be argued that quality, professional grade mike inputs *do* have an input transformer and therefore one side may be grounded with impunity. This euphoria is quickly dispelled by the realization that when a real microphone is connected to the input, the system is then *balanced* so that operating in an unbalanced mode is not representative of things as they really are. To paraphrase Walter Cronkite's famous tag line: that isn't the way it is.

The second difficulty encountered in the conventional approach is getting the proper signal level. One audio oscillator has a step attenuator with 10 dB per step, but most of the "garden variety" of oscillators have nothing more sophisticated than a simple pot— and probably a linear taper to boot! With such a control it is virtually impossible to attain a miniscule output voltage on the order of one millivolt. The harried engineer is between Scylla and Charybdis: on one hand is zero signal and then, with the slightest rotation, the amplifier is overloaded. Then there is the ever-present problem of drift, especially with carbon controls, for after carefully finding the correct signal level, one finds that the signal has now increased or decreased by a hair's breadth and another round of setting comes up. Never forget that the inexorable decibel laws operate with minute signals in exactly the same manner as with larger ones. Thus, going from one millivolt to two (as by control drift) represents a 6 dB change in level, and this error is built into the measurement!



ITEM	DESCRIPTION	QTY
1	Dual binding posts, ¾" spacing, Red-Black with mounting base	1
2	Microphone connector, 3 pin male	1
3	Transformer, 600/600Ω, one winding split and strapped for 150Ω	1
4	Aluminum mini box enclosure	1
5	Resistor, fixed carbon, 0.5 watt, 75Ω, ±5%	2
6	Resistor, fixed, carbon, 0.5 watt, 160Ω, ±5%	2
7	Resistor, fixed, carbon, 0.5 watt, 130Ω, ±5%	2
8	Resistor, fixed, carbon, 0.5 watt, 200Ω, ±5%	2
9	Resistor, fixed, carbon, 0.5 watt, 110Ω, ±5%	2
10	Resistor, fixed, carbon, 0.5 watt, 150Ω, ±5%	1
11	Resistor, fixed, carbon, 0.5 watt, 36Ω, ±5%	2

Figure 1. The three parts of the artificial microphone: a transformer, a resistive attenuator and a build-out network which restores the 150 ohm balanced impedance.

IMPEDANCE MATCHING

In recent years the principle of impedance matching has fallen into disrepute—and this is, perhaps, unfortunate. Be that as it may, there is an area associated with Proof of Performance measurement in which impedance matching is important. All well designed (in this author's opinion!) microphone amplifiers are equipped with an input transformer with a primary impedance of 150 ohms. When a transformer designer assigns numerical values to the impedances of the windings, it does *not* mean that these are the actual impedances of the windings. Rather, they are the values of source and load impedance between which the transformer is designed, and intended, to work. Thus, for example, a microphone input transformer with a 150-ohm primary and a 3,000-ohm secondary impedance rating, means that the transformer should be fed from a 150-ohm Thevenin source and, for the purpose of measurement, terminated with a 3,000-ohm resistor. Since the output impedance (Thevenin source impedance) of many audio oscillators is 600 ohms, it immediately follows that this is four times the correct impedance from which a 150-ohm microphone transformer should be fed. Feeding a transformer from a source impedance greater than the design value has two deleterious effects. As is well known, the transformer's low frequency response begins to fall off when the primary inductive reactance equals the source impedance. Thus for a higher source resistance, the −3 dB rolloff begins at a higher frequency. The second effect is, perhaps, not so well known. When a transformer is connected to a source of AC, a current is drawn by the transformer which sets up the magnetic flux in the core. This current is called the "exciting current" and is non-linear. If the signal source is a sine wave and the transformer is fed through a source resistance, the distorted exciting current will produce a voltage drop across the source resistance which adds vectorially to the sine wave signal, producing a distorted voltage which is impressed on the transformer primary and thus appears on the secondary. To avoid this problem, a transformer should always be fed from its rated source impedance, *never* from a higher impedance, such as feeding a 150-ohm primary from a 600-ohm (four times) source. Exciting current distortion is well known to transformer designers and references to it may be found in transformer texts such as "Magnetic Circuits & Transformers" by the EE Staff of Massachusetts Institute of Technology, John Wiley, 1943, pages 160-173.

THE ARTIFICIAL MICROPHONE

The artificial microphone is a device which electrically simulates a real microphone in signal level, balanced circuitry and source impedance. It makes possible accurate and repeatable measurements in microphone circuits including amplitude-frequency response, distortion and residual noise.

As shown in FIGURE 1, the device is, like Caesar's Gaul, divided into three parts. These are (1) a transformer; (2) a resistive attenuator; (3) a build-out network which restores the 150 ohm balanced impedance.

The transformer is used in the high-level portion of the circuit for two reasons: (1) with a relatively high level of signal, the requirements of shielding are much less severe; (2) the performance of many transformers is degraded with very low values of input signal of 1 millivolt or less. The transformer used is a repeat coil with unity turns ratio and with one winding split. This winding has the sections connected in parallel rather than in series with a consequent impedance of 150 ohms. The transformer must be of the highest quality that

the budget can afford since it is the sole element which determines the amplitude-frequency response of the artificial microphone.

The second section consists of two lattice attenuators connected in series. The lattice network is an excellent attenuator for balanced circuits. Two sections are used—one of 24 dB and the other of 16 dB—for a total loss of 40 dB, or a voltage attentuation of 100 times. The resistor values have been slightly modified to fit the 5 percent preferred values which are more readily available than the "funny" values turned up by pad formulas. Two sections are used because, if one attempts to design a single section of 40 dB loss or more, the values come out so close to one another that resistors must be individually measured on a bridge. It's the old story of the difference between two large quantitites. Two sections are much better.

The third section has two series resistors which restore the 150 ohm source resistance needed. It will be recalled that the impedance of a terminated pad is half of the termination.

The two resistors immediately following the transformer are used to provide a 150 ohm source resistance for the pad, since the signal at the input terminals is held constant during a frequency run and therefore is a zero impedance point. These resistors also provide a 6 dB voltage attenuation at no extra cost.

The overall device has an attenuation, including the transformer, of 52 dB or a voltage attenuation of 398.11 times or a ratio of 0.00251. Thus, if 0.5 volt is applied at the input terminals, the open circuit output voltage will be 0.00126 volt. This theoretical value will be subject to some variation due to the tolerance variation in the resistors. A unit constructed with this circuit measured almost exactly one millivolt out for 0.5 volt in. Since one millivolt behind 150 ohms represents an available power of -57.8 dBm, the signal from the artificial microphone simulates the real thing. The circuit is passive and linear so that signals at the output of other power levels are readily obtainable by varying the input signal. The output will vary in a linear manner, dB for dB.

The author's unit was constructed in an aluminum "Mini-Box" with the transformer inside. The pads assemble very readily on a turret type insulating board. The input connector is a pair of binding posts with ¾-inch spacing. The output is a standard three-pin microphone.

Since there is a resistive termination at the output and some 52 dB attenuation toward the input, a system's residual noise level may be measured by simply removing the signal source and terminating the input with a double banana plug with a 620 ohm resistor between the two plugs.

A few tips on checking a completed unit may be in order. The amplitude-frequency response of the transformer may be measured easily if one has a voltage that is constant with frequency behind a 600 ohm source. Such is the case with most audio oscillators from the major manufacturers. If one is not sure of the output impedance it may be readily measured. Simply measure the open circuit (no load) voltage from the oscillator. Then connect a 620 ohm resistor across the generator's output and remeasure the voltage. If the output (source) impedance is 600 ohms, the voltage will be halved (6 dB drop). The response is measured with a voltmeter connected across the transformer's secondary. Don't worry about balance here; the series resistors are further along in the circuit.

To measure the output voltage (open circuit), one *must* consider balance. The sensitive electronic voltmeters capable of measuring one millivolt are invariably *un*balanced. Thus we use a 1:1 transformer or repeat coil ahead of the voltmeter. Needless to say, a 'scope is mighty handy when checking these minute voltages. It will quickly tell whether one is measuring a 1,000 Hz tone or some stray hum that sneaked in when you weren't listening! Electronic voltmeters have an output connection using the meter amplifier and these are a convenient place for a 'scope connection. ■

New Products & Services

REMOTE CONTROLLED AUDIO ATTENUATOR

• FSR, Inc. has recently introduced a new remote audio attenuator specifically designed for high quality sound systems. Available in single (LDR-1) or dual (LDR-2) channel, the units feature a 12 dB pad bypass already mounted on pc board; no audio distortion and no control induced noise. The audio attenuators operate off 24VDC power.
Mfr: FSR, Inc.
Circle 46 on Reader Service Card

TAPE TRANSPORT BRAKING SYSTEM

• Pentagon Industries, Inc. has developed a new concept for reel to reel tape transport braking system that is particularly applicable to high speed duplication of audio tapes. The compound electro-mechanical brake system, currently employed in the 1100 Series Duplicating System, provides smooth handling for all standard tapes and has enabled open reel duplication at speeds of 120 inches per second. It does not require adjustments due to wear or temperature and humidity changes. Four inch reel masters in connection with "Nagra SN" recordings of .150 mil wide times one quarter mil thick master tapes are duplicated without any danger of stretching, as in the case of conventional brakes.
Mfr: Pentagon Industries, Inc.
Circle 48 on Reader Service Card

ANALOG BARGRAPH

• The APM Series 20 is a new solid-state analog bargraph indicator featuring a 3-in., 20 element bargraph. The APM 20 meters provide a low cost alternative to mechanical meters and are available in a number of standard voltmeter and ammeter ranges. Options include offset span, differential input, reduced response time and either single or dual setpoint controls. Meters meet ANSI 39.1 shock, temperature and humidity requirements. Display brightness is controllable with an external potentiometer.
Mfr: Bowmar/ALI Inc.
Price: $60.00 and down
Circle 47 on Reader Service Card

RMS ANALOG VOLTMETER

• The IET Model MV-800 Analog Voltmeter is a true RMS instrument and will therefore accurately present the effective power of waveforms which depart from a true sine wave. Fifteen ranges are provided from 30μV full scale to 300V, and from -90 dB full scale to +50 dB in decades. A front panel bandwidth switch permits a choice of wide band measurements, standard first audio bandpass (20-20 kHz) or external filter. A group of standard or custom plug-in filter modules are available from IET. Internal rechargeable nicad batteries allow the Model MV-800 to be totally isolated from the power line and external grounds. The Model MV-800 can be used as an oscilloscope preamplifier through the use of its front panel BNC output jack. A chart recorder, DVM, X-Y plotter may be connected to the rear BNC connector for permanent records of readings or under-over limits observations. This terminal provides a linear DC output of 0-1 volt proportional to meter deflection.

Mfr: IET Labs, Inc.

Price: $495.00

Circle 49 on Reader Service Card

SYNTHESIZER

• The Realistic MG-1 Synthesizer by Moog is a musical synthesizer with professional features designed for club or concert performance, composing and rehearsing. The MG-1 features a 2½-octave full-chromatic keyboad and a versatile control panel, divided into related sections and color-coded for easy use. Two independent three-octave tone sources feature variable waveshapes; Tone Source One offers a 2:1 sync selector, 3-position octave selector and 2-position (square/triangular) waveshape selector. Tone Source Two offers a detune control for dissonant or full-interval offsetting of its pitch, 3-position octave selector and 2-position waveshape selector. The modulation section can create vibrato and tremolo effects, pulsating notes, random tone sequences and glide effects. It controls the tone sources, filter, keyboard glide and modulation rate, and includes a 3-position waveshape selector and 2-position auto-contour trigger. The 4-control filter adjusts the tonal characteristics (timbre) of the synthesizer's output by controlling cutoff frequency, peak emphasis and contoured cutoff. A master volume control adjusts the output of the synthesizer, which is available both at a headphone jack and at phonotype output jacks that permit direct connection for most stereo and sound systems.

Mfr: Tandy Corporation/Radio Shack

Circle 50 on Reader Service Card

PRO MIXING CONSOLE

• The Series 1000 professional mixing console is made for small sound reinforcement jobs, submixers, keyboard mixers, as well as for theatrical, church and broadcasting applications. They are available in 8, 12 and 16 channel versions (the 8 channel is available in a 19-in. rack-mounted version). The Series 1000 has balanced transformerless inputs and outputs, and accepts high and low impedance inputs. Input attenuation switches allow precise and repeatable selection of input gain from +50 dB to −10 dB in 10 dB steps. Each channel has a LED preamp level monitor which indicates critical headroom level, and has a responsive 3-band EQ. Other standard features include: output patch points for inserting EQ's, limiters, 10-segment LED bar graph displays, headphone monitoring, and work lamp socket. A sum switch converts the stereo sends into subgroups. The built-in reverb system is patchable for external effects, using the "Audy effects loop" concept.
Mfr: Audy Instruments, Inc.
Circle 51 on Reader Service Card

SILENT PLUGS AND PHONE PLUGS

• Switchcraft Silent Plugs feature a circuit closing device that stops hum, squeals and pops when plug is removed from the jack. Also featured are new solder lug terminals and mylar tube insert for greater electrical protection, and one-piece tip rod assembly to insure plug quality. Silent Plugs utilize cables up to .25-in. maximum diameter and can accommodate parallel or shielded cable. The phone plugs also employ new solder lug terminals for easy connection, with one piece tip rod standard on all plugs. Four different models are available.
Mfr: Switchcraft, Inc.
Circle 52 on Reader Service Card

REMOTE SWITCHED ISOLATORS

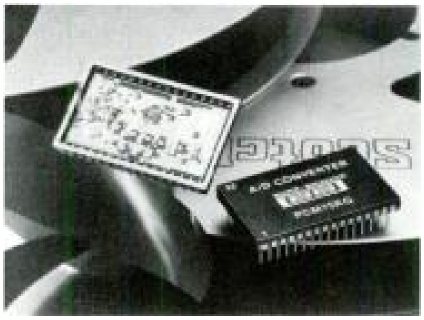
• Remotely Switched Isolators announced by Electronic Specialists are the newest additions to their patented isolator filter/suppressor line of interference control products. Available on all isolator models, the remote AC power control switch can be mounted with the audio equipment for convenience. Total Remote Switched Isolator capacity is 1875 watts max, with each socket capable of handling a 1 KW load.
Mfr: Electronic Specialists, Inc.
Price: $79.95 and up
Circle 53 on Reader Service Card

FLOOR MONITOR

• The TA-12 is a compact floor monitor suited for applications ranging from rock 'n roll to speech or acoustical instrument reinforcement. The system utilizes an E-12 Bag End™ 12-in. loudspeaker and an ST350B Electro-Voice tweeter in a specially designed crossover network with solid state tweeter protection. The system employs the Time Alignment Technique™ of E. M. Long Associates, making it the first Time Align performance monitor of its kind.
Mfr: Modular Sound Systems Inc.
Circle 54 on Reader Service Card

ANALOG-TO-DIGITAL CONVERTER

• The PCM75 A/D converter is designed for PCM Audio applications and is compatible with EIAJ STC-007 specifications. The internal 16-bit digital-to-analog converter is available for the designer to utilize in the playback mode. The conversion time can be reduced from 15 µsec to 8 µsec with some increase in distortion. Distortion is specified on the data sheet to assure performance in critical audio applications. The PCM75 is complete with internal reference and clock.
Mfr: Burr-Brown Research Corporation
Circle 55 on Reader Service Card



PATCH CORD

• The Model 4812 Minigrabber patch cord utilizes 22 AWG, two conductor shielded microphone cable, and standard PJ-051 R phone plug. The gold plated, beryllium copper Minigrabber provides maximum sensitivity.
Mfr: ITT Pomona Electronics
Price: $19.50
Circle 56 on Reader Service Card

SIGNAL MULTIPLEXER

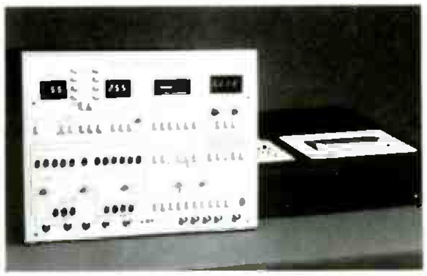
• The 1360 Programmable Signal Multiplexer is a microprocessor-based, IEEE-488 compatible, system instrument that can be used to multiplex electrical signals. The 1360 includes two separate chassis: the 1360P Programmable Switch Controller and the 1360S Switch Matrix. Up to four 1360S Switch units can be operated by the same 1360P Controller. For each Switch Matrix, the user can choose to multiplex one output with thirty-two inputs, two ganged outputs with sixteen inputs or four ganged outputs with eight inputs.
Mfr: Tektronix, Inc.
Price: 1360P—$2,500.00,
1360S—$1,500.00
Circle 57 on Reader Service Card

Circle 13 on Reader Service Card

ROTARY SWITCH

• The Model RT-A is a printed circuit mounting rotary switch for use in test equipment, oscilloscopes and medical equipment. The Model RT-A, with binary or decimal coding, features moveable contacts that are obtained by coded discs and housed in a flat case. The contacts, BBM or MBB, are silver plated and have 5 VA switching power. RT-A contacts are designed to be protected against flux contamination at the moment of soldering (in the bath or flow) and printed circuit board cleaning. Available options include a threaded bushing mount, adjustable stop, screwdriver slot, flatted shaft, solder lugs or P.C. pins.
Mfr: ITT Schadow, Inc.
Price: 1K Pieces, $1.39
Circle 58 on Reader Service Card

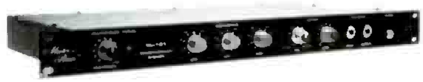
AUDIO SPECTRUM ANALYZER



• The Digital Sona-Graph (Model 7800) is an audio spectrograph which combines the features of a conventional spectrum analyzer, a 3-D display, a high resolution grey scale printer and an oscillograph. The signal is recorded in a 64K word by 10 bit memory. As much as 2.56 seconds of signals at 8 kHz (up to 41 seconds at 500 Hz) can be analyzed using several display modes and filter bandwidths. Frequency changes over time can be displayed and analyzed. As little as 6ms of any portion of the time domain signal can be displayed and printed.
Mfr: Kay Elemetrics Corp.
Price: $13,500.00
Circle 59 on Reader Service Card

Circle 41 on Reader Service Card

REVERBERATION SYSTEM



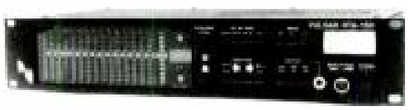
• The XL-121 Reverberation System is designed to interface with virtually any other audio equipment. The Preamp Gain control allows the unit to accept a low-level musical instrument output such as an electric guitar, as well as higher level signals associated with sound reinforcement consoles. In addition, it can serve as a preamplifier to interface directly with any power amplifier. This feature enables the reverb to be used in the signal chain at all times, and a standard footswitch can be used to switch the reverb effect in and out without affecting the direct signal. The Output Level control also permits interfacing with all other signal processing equipment. The front panel Output Mix control allows blending of any desired amount of reverberated and direct signal. The Equalization section is intended to allow the user to tailor the reverberant sound characteristics. Included in the section is a Low, Mid and High control, all with 12 dB of boost or cut. The EQ section only affects the sound of the reverb, and the controls are non-interacting. A dual-colored LED serves as a Power Overload indicator.
Mfr: MICMIX Audio Products, Inc.
Price: $450.00
Circle 60 on Reader Service Card

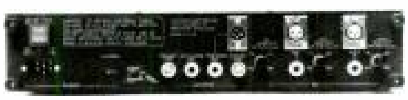
SOUND SYSTEM

• The Entertainer is a complete, portable, powered sound system consisting of a 10-channel 150 watt per channel powered mixing console and two constant directivity speaker systems. The 100M powered mixing section of the system has a ten-input stereo out configuration. Two channels have line-level inputs that can be used with tape decks, allowing pre-recorded portions of a performance to be introduced. The 100M also features fluroscan bar graph displays for across-the-stage meter reading. The 100S speaker system is housed in a polyethylene cabinet. The system's twelve-inch woofer is complemented by a constant directivity tweeter. The 100S systems are rated to handle 100 watts long-term and up to 400 watts for short-term peaks.
Mfr: E-V/TAPCO
Price: $2,195.00
Circle 61 on Reader Service Card

REAL TIME ANALYZER

• The RTA 150 real time analyzer is a combination of four instruments in one: real time analyzer, sound level meter, wave analyzer and pink noise generator. The RTA 150 has a matrix of 209 LEDs with 36 dB of displayed dynamic range. An over or under range arrow indicates the switch to press to get the spectrum back on display. In addition to 15 frequency bands, there is an extra yellow band that continuously displays the broad band sound pressure level (SPL) of all frequencies. The yellow horizontals LEDs become the flat reference line. The digital display that corresponds to the LEDs can be adjusted from 50 dB to 125 dB (in 5 dB increments) by pressing the reference buttons up or down as needed. The RTA 150 uses a scan function that allows you to place a bright display cursor at any desired frequency band. The display of the scanned frequency will appear much brighter and the decay rate will increase to follow signal transients more accurately. The scanned frequency will also appear on the analog display meter to allow that band to be read in 1 dB increments. A 31-bit pseudorandom digital number generator provides a stable, accurate noise source. The pink noise buttons, located on the front panel, are adjustable from 80 to 110 dB in 10 dB increments. When the pink noise generator is switched off, the gain of the pink noise automatically returns to the 80 dB gain level.
Mfr: Pulsar Laboratories, Inc.
Circle 63 on Reader Service Card





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db People/Places/Happenings

• **Rupert Neve Incorporated** has announced the appointment of Mr. **Anthony H. Langley** as vice president sales. Langley was first employed by Neve's parent company in England starting in 1972 in product design and production engineering. He was appointed to manager of Test and Quality Control at the Kelso, Scotland factory in 1974, which position he held until early 1976 when he was transferred to Rupert Neve Incorporated as marketing support engineer. Since 1977 Langley has held the position of marketing manager for Neve's U.S. operation.

• **Joe Bean** has joined **Studer Revox America** as a sales representative, according to the firm's president, **Bruno Hochstrasser.** Bean will concentrate on developing the broadcast market for the Studer professional lines in the Southeast. Bean comes to Studer after serving for 3½ years as a sales representative for **Audio Consultants,** a Nashville studio supply and design company.

• The Denver-based acoustical consulting firm of **David L. Adams Associates** has announced a relocation of its offices to larger quarters at 1701 Boulder Street, Denver, Colorado 80211. A new brochure describing the firm's experience and qualifications is also available upon request.

• **Bose Corporation** has announced that **John Strand** has been appointed to the post of applications engineer, Professional Products. Strand, a native of Milwaukee, is a graduate of the University of Wisconsin in Electrical Engineering and Acoustics. He will be based at Bose corporate headquarters in Framingham, Massachusetts. Strand will head Bose's liaison program with acoustical consultants and sound system designers and will provide applications assistance to sound contractors and end users of Bose Professional Products. The appointment of Strand, according to Bose Pro Marketing Manager **Roy Komack,** marks a new phase in Bose Corporation's expansion in the engineered sound business. This program will take its next step with a major new product introduction this coming winter.

• **Digital Recording Corporation** (NASDAQ-DRSO) announced the opening of its new Nashville facility for its **Soundstream, Inc.** subsidiary. The new facility will be equipped with Soundstream's advanced digital recording equipment for serving the Midwest and Southeast recording markets. A demonstration of Soundstream equipment was held on July 20 in Nashville to which the trade press, recording industry, and the public was invited.

• **Bob Silk** and **Paul Blank,** owners of **The Mike Shop,** are pleased to announce that The Mike Shop has moved to new, larger quarters in Lynbrook, New York in order to provide improved sales and service to its customers. Comprising 4000 square feet, the newly renovated building will house The Mike Shop's offices, service department, shipping department, and warehouse space, all on one floor. Originally offering only microphones and accessories, The Mike Shop now specializes in providing a full line of professional audio equipment and services to recording studios, broadcast stations, churches and independent sound engineers. The Mike Shop's mailing address and telephone number have not been changed.

• The Motown Sound is now being exclusively recorded and mixed on **U.R.E.I**'s new Series "A" Time-Aligned studio monitoring professional loudspeakers. Installed as a standard reference system throughout the **Motown/Hitsville** multi-studio complex (5), the first pair of 811's retrofitted into the control room soffit without extensive modifications.

• **Paul Murphy,** general manager of **Beyer Dynamic, Inc.,** announced the appointment of Tom **Bensen** as national sales manager for Consumer and Professional Products. Bensen has a varied background in the electronics industry which includes 2½ years as Audio Products division manager for **Eumig.** He was technical director for **TDK** before joining Eumig and was announcer/engineer for **Beck-Ross Communications Inc.** (WGLI Radio) for 2 years.

• **Mr. Hosoda,** president of **Otari Electric Co., Ltd.,** has announced the official opening of Otari Electric Deutschland GMBH, (West Germany). The sales and service organization, located near Dusseldorf, is headed up by **Mr. Ken Hirano,** operations manager. Opened in June, the new branch of the Japanese professional audio tape machine manufacturer will handle European business and service for all Otari products.

• **Tentel,** manufacturer of the Tentelometer tape tension gage and similar service tooling, has announced a move into newer and larger quarters. Their new address is 1506 Dell Avenue, Campbell, California 95008. The new phone number is (408) 379-1881 or toll-free (800) 538-6894, continental U.S. except California.

• **Bonneville Productions** has installed a Q-Lock 310 **SMPTE** time code synchronizer system. The system is portable and can be used in any of Bonneville's three specialized studios to run two or three tape sources simultaneously in exact synchronization. It is interfaced to **Ampex** brand MM1200 twenty-four track, 440C eight track, ATR104 four track audio machines and to a **Sony** 2860 U-matic video cassette recorder.

• **Hal** and **Vio Michael** have recently announced the completion of **Spindletop Recording Studios** in Hollywood, CA. Spindletop features an **MCI** 636 console, complete with JH-50 automation, JH-24 24-track recorders, JH-110B ¼-in. 2-track recorder, JH-110B ½-in. 4-track recorder, JH-110B ½-in. 2-track mastering recorder, and a JH-45 SMPTE-EBU Based Generator/Synchronizer, providing the capability of 48-track audio as well as video interlock.

• The appointment of **Dom Notto** as general manger/vice president of **Nagra Magnetic Recorders, Inc.,** was announced recently by **Kudelski, S.A.** of Switzerland, parent group for NMRI. Mr. Notto who, for 10 years, has been vice president, sales for the company, succeeds **Jean-Jacques Broccard,** who remains general manager of NMRI of California.


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A & R Recording ___ Air Studios ___ Alpha International Recording Studio, Inc. ___ Ardent Recording, Inc. ___ Bayshore Recording Studio ___ Cherokee Studios ___ Columbia Studios
Audio ___ Apogee Studio ___ Applewood Studios ___ Backstage Studios ___ Celebration Recording ___ Cybertechnicks ___ Davlen Sound Studio
The Automatt ___ Axis Studios ___ Capitol Records ___ Different Fur Studios ___ Dimensional Echo Sound
CBS Studios ___ Can-Base Productions ___ Group IV Recording, Inc. ___ The Hit Factory
Criteria Recording Studio ___ Wally Heider Studios ___ Jack's Tracks Recording Studio
Delphine Studios ___ Kendun Recorders ___ Konk Studio
Dimension Sound Studios ___ Le Studio, Morin Heights
The Enactron Truck ___ Love 'n' Comfort ___ MCA
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House of Music, Inc. ___ Master Disc ___ The Mixing Lab ___ Muscle
Kaye-Smith Productions ___ North American Recording
Las Vegas Recording Studio ___ P.S. Recording Studio
Longview Farms ___ Paramount Recording
MCA/Whitney Studio ___ Plaza Sound Studios ___ Polar
The Middle Ear ___ Minot Sound ___ Pumpkin Records
Muscle Shoals Sound Studio ___ Quadrafonic Sound Studio
Nova Studio ___ One Step Up ___ Recording Studio MEDI
Pacific Recording Studio ___ Remote Truck ___ Ridge F
Phase One Studios ___ Rusk Sound Studio
Polar Music Studio ___ Power Station ___ Sea-West Studios
Quadradial ___ The Shelter Studio
Record Plant ___ Sound City Studios
Reel Sound Recording Co. ___ Sound Labs
Ridge Farm Studio ___ Sound Recorder Studios
S.S.C. Studio ___ Sarm Studio ___ Sundance
Secret Sound Studio ___ Studio Sound Recorders ___ United S
Shoe Productions ___ Studio One ___ Ultra Sonic Recording Studios
Sound Ideas Recording Studios ___ Studio Masters ___ The Village Recorder ___ Warner
Sound Pit Studios ___ Studio in the Country ___ Trolley Track Sound Studios ___ Victor Studio ___ A & R Record
Sound Palace ___ Studio Acousti ___ The Townhouse ___ Universal Recording Studios ___ Wishbone Studio ___ American Studio
Soundworks ___ Superdisc ___ United Western ___ Windmill Lane ___ Ameraycan Studios
Sundance Productions, Inc. ___ Westlake Audio ___ Dale Ashby and Father ___ Atlantic Studios ___ Bearsville
United Sound Systems, Inc. ___ Wessex Recording Studio ___ Alpha International Recording Studio, Inc. ___ Bayshore Recording Studio ___ Can-Base Prod
Warner-Pioneer Recording Studio ___ Alpha Audio ___ Ardent Recording, Inc. ___ Backstage Studios ___ CBS Studios ___ Cherokee
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